INSTALLATION AND OPERATION MANUAL

# VC-E1/16, VC-E1/12

## Voice Compression Modules

Gmux-2000 Ver. 3.2

RAD
data communications
The Access Company

# VC-E1/16, VC-E1/12

Voice Compression Modules

Gmux-2000 Ver. 3.2

Installation and Operation Manual

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# Limited Warranty

RAD warrants to DISTRIBUTOR that the hardware in the VC-E1/16, VC-E1/12 to be delivered hereunder shall be free of defects in material and workmanship under normal use and service for a period of twelve (12) months following the date of shipment to DISTRIBUTOR.

If, during the warranty period, any component part of the equipment becomes defective by reason of material or workmanship, and DISTRIBUTOR immediately notifies RAD of such defect, RAD shall have the option to choose the appropriate corrective action: a) supply a replacement part, or b) request return of equipment to its plant for repair, or c) perform necessary repair at the equipment's location. In the event that RAD requests the return of equipment, each party shall pay one-way shipping costs.

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This Agreement shall be construed and governed in accordance with the laws of the State of Israel.

# Product Disposal

To facilitate the reuse, recycling and other forms of recovery of waste equipment in protecting the environment, the owner of this RAD product is required to refrain from disposing of this product as unsorted municipal waste at the end of its life cycle. Upon termination of the unit's use, customers should provide for its collection for reuse, recycling or other form of environmentally conscientious disposal.

# General Safety Instructions

The following instructions serve as a general guide for the safe installation and operation of telecommunications products. Additional instructions, if applicable, are included inside the manual.

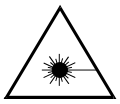
# Safety Symbols

***Warning***

This symbol may appear on the equipment or in the text. It indicates potential safety hazards regarding product operation or maintenance to operator or service personnel.

Danger of electric shock! Avoid any contact with the marked surface while the product is energized or connected to outdoor telecommunication lines.

Protective ground: the marked lug or terminal should be connected to the building protective ground bus.



***Warning***

Some products may be equipped with a laser diode. In such cases, a label with the laser class and other warnings as applicable will be attached near the optical transmitter. The laser warning symbol may be also attached.

Please observe the following precautions:

- Before turning on the equipment, make sure that the fiber optic cable is intact and is connected to the transmitter.
- Do not attempt to adjust the laser drive current.
- Do not use broken or unterminated fiber-optic cables/connectors or look straight at the laser beam.
- The use of optical devices with the equipment will increase eye hazard.
- Use of controls, adjustments or performing procedures other than those specified herein, may result in hazardous radiation exposure.

ATTENTION: The laser beam may be invisible!

In some cases, the users may insert their own SFP laser transceivers into the product. Users are alerted that RAD cannot be held responsible for any damage that may result if non-compliant transceivers are used. In particular, users are warned to use only agency approved products that comply with the local laser safety regulations for Class 1 laser products.

Always observe standard safety precautions during installation, operation and maintenance of this product. Only qualified and authorized service personnel should carry out adjustment, maintenance or repairs to this product. No installation, adjustment, maintenance or repairs should be performed by either the operator or the user.

# Handling Energized Products

## General Safety Practices

Do not touch or tamper with the power supply when the power cord is connected. Line voltages may be present inside certain products even when the power switch (if installed) is in the OFF position or a fuse is blown. For DC-powered products, although the voltages levels are usually not hazardous, energy hazards may still exist.

Before working on equipment connected to power lines or telecommunication lines, remove jewelry or any other metallic object that may come into contact with energized parts.

Unless otherwise specified, all products are intended to be grounded during normal use. Grounding is provided by connecting the mains plug to a wall socket with a protective ground terminal. If a ground lug is provided on the product, it should be connected to the protective ground at all times, by a wire with a diameter of 18 AWG or wider. Rack-mounted equipment should be mounted only in grounded racks and cabinets.

Always make the ground connection first and disconnect it last. Do not connect telecommunication cables to ungrounded equipment. Make sure that all other cables are disconnected before disconnecting the ground.

Some products may have panels secured by thumbscrews with a slotted head. These panels may cover hazardous circuits or parts, such as power supplies. These thumbscrews should therefore always be tightened securely with a screwdriver after both initial installation and subsequent access to the panels.

## Connecting AC Mains

Make sure that the electrical installation complies with local codes.

Always connect the AC plug to a wall socket with a protective ground.

The maximum permissible current capability of the branch distribution circuit that supplies power to the product is 16A. The circuit breaker in the building installation should have high breaking capacity and must operate at short-circuit current exceeding 35A.

Always connect the power cord first to the equipment and then to the wall socket. If a power switch is provided in the equipment, set it to the OFF position. If the power cord cannot be readily disconnected in case of emergency, make sure that a readily accessible circuit breaker or emergency switch is installed in the building installation.

In cases when the power distribution system is IT type, the switch must disconnect both poles simultaneously.

## Connecting DC Power

Unless otherwise specified in the manual, the DC input to the equipment is floating in reference to the ground. Any single pole can be externally grounded.

Due to the high current capability of DC power systems, care should be taken when connecting the DC supply to avoid short-circuits and fire hazards.

DC units should be installed in a restricted access area, i.e. an area where access is authorized only to qualified service and maintenance personnel.

Make sure that the DC power supply is electrically isolated from any AC source and that the installation complies with the local codes.

The maximum permissible current capability of the branch distribution circuit that supplies power to the product is 16A. The circuit breaker in the building installation should have high breaking capacity and must operate at short-circuit current exceeding 35A.

Before connecting the DC supply wires, ensure that power is removed from the DC circuit. Locate the circuit breaker of the panel board that services the equipment and switch it to the OFF position. When connecting the DC supply wires, first connect the ground wire to the corresponding terminal, then the positive pole and last the negative pole. Switch the circuit breaker back to the ON position.

A readily accessible disconnect device that is suitably rated and approved should be incorporated in the building installation.

If the DC power supply is floating, the switch must disconnect both poles simultaneously.

# Connecting Data and Telecommunications Cables

Data and telecommunication interfaces are classified according to their safety status.

The following table lists the status of several standard interfaces. If the status of a given port differs from the standard one, a notice will be given in the manual.

| Ports | Safety Status |
|---|---|
| V.11, V.28, V.35, V.36, RS-530, X.21, 10 BaseT, 100 BaseT, Unbalanced E1, E2, E3, STM, DS-2, DS-3, S-Interface ISDN, Analog voice E&M | SELV Safety Extra Low Voltage:<br>Ports which do not present a safety hazard. Usually up to 30 VAC or 60 VDC. |
| xDSL (without feeding voltage), Balanced E1, T1, Sub E1/T1 | TNV-1 Telecommunication Network Voltage-1:<br>Ports whose normal operating voltage is within the limits of SELV, on which overvoltages from telecommunications networks are possible. |
| FXS (Foreign Exchange Subscriber) | TNV-2 Telecommunication Network Voltage-2:<br>Ports whose normal operating voltage exceeds the limits of SELV (usually up to 120 VDC or telephone ringing voltages), on which overvoltages from telecommunication networks are not possible. These ports are not permitted to be directly connected to external telephone and data lines. |
| FXO (Foreign Exchange Office), xDSL (with feeding voltage), U-Interface ISDN | TNV-3 Telecommunication Network Voltage-3:<br>Ports whose normal operating voltage exceeds the limits of SELV (usually up to 120 VDC or telephone ringing voltages), on which overvoltages from telecommunication networks are possible. |

**Always connect a given port to a port of the same safety status. If in doubt, seek the assistance of a qualified safety engineer.**

Always make sure that the equipment is grounded before connecting telecommunication cables. Do not disconnect the ground connection before disconnecting all telecommunications cables.

Some SELV and non-SELV circuits use the same connectors. Use caution when connecting cables. Extra caution should be exercised during thunderstorms.

When using shielded or coaxial cables, verify that there is a good ground connection at both ends. The grounding and bonding of the ground connections should comply with the local codes.

The telecommunication wiring in the building may be damaged or present a fire hazard in case of contact between exposed external wires and the AC power lines. In order to reduce the risk, there are restrictions on the diameter of wires in the telecom cables, between the equipment and the mating connectors.

***Caution*** To reduce the risk of fire, use only No. 26 AWG or larger telecommunication line cords.

***Attention*** Pour réduire les risques s'incendie, utiliser seulement des conducteurs de télécommunications 26 AWG ou de section supérieure.

Some ports are suitable for connection to intra-building or non-exposed wiring or cabling only. In such cases, a notice will be given in the installation instructions.

Do not attempt to tamper with any carrier-provided equipment or connection hardware.

# Electromagnetic Compatibility (EMC)

The equipment is designed and approved to comply with the electromagnetic regulations of major regulatory bodies. The following instructions may enhance the performance of the equipment and will provide better protection against excessive emission and better immunity against disturbances.

A good ground connection is essential. When installing the equipment in a rack, make sure to remove all traces of paint from the mounting points. Use suitable lock-washers and torque. If an external grounding lug is provided, connect it to the ground bus using braided wire as short as possible.

The equipment is designed to comply with EMC requirements when connecting it with unshielded twisted pair (UTP) cables. However, the use of shielded wires is always recommended, especially for high-rate data. In some cases, when unshielded wires are used, ferrite cores should be installed on certain cables. In such cases, special instructions are provided in the manual.

Disconnect all wires which are not in permanent use, such as cables used for one-time configuration.

The compliance of the equipment with the regulations for conducted emission on the data lines is dependent on the cable quality. The emission is tested for UTP with 80 dB longitudinal conversion loss (LCL).

Unless otherwise specified or described in the manual, TNV-1 and TNV-3 ports provide secondary protection against surges on the data lines. Primary protectors should be provided in the building installation.

The equipment is designed to provide adequate protection against electro-static discharge (ESD). However, it is good working practice to use caution when connecting cables terminated with plastic connectors (without a grounded metal hood, such as flat cables) to sensitive data lines. Before connecting such cables, discharge yourself by touching ground or wear an ESD preventive wrist strap.

# FCC-15 User Information

This equipment has been tested and found to comply with the limits of the Class A digital device, pursuant to Part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the Installation and Operation manual, may cause harmful interference to the radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

# Canadian Emission Requirements

This Class A digital apparatus meets all the requirements of the Canadian Interference-Causing Equipment Regulation.

Cet appareil numérique de la classe A respecte toutes les exigences du Règlement sur le matériel brouilleur du Canada.

# Warning per EN 55022 (CISPR-22)

| | |
|---|---|
| ***Warning*** | This is a class A product. In a domestic environment, this product may cause radio interference, in which case the user will be required to take adequate measures. |
| ***Avertissement*** | Cet appareil est un appareil de Classe A. Dans un environnement résidentiel, cet appareil peut provoquer des brouillages radioélectriques. Dans ces cas, il peut être demandé à l'utilisateur de prendre les mesures appropriées. |
| ***Achtung*** | Das vorliegende Gerät fällt unter die Funkstörgrenzwertklasse A. In Wohngebieten können beim Betrieb dieses Gerätes Rundfunkströrungen auftreten, für deren Behebung der Benutzer verantwortlich ist. |

# Mise au rebut du produit

Afin de faciliter la réutilisation, le recyclage ainsi que d'autres formes de récupération d'équipement mis au rebut dans le cadre de la protection de l'environnement, il est demandé au propriétaire de ce produit RAD de ne pas mettre ce dernier au rebut en tant que déchet municipal non trié, une fois que le produit est arrivé en fin de cycle de vie. Le client devrait proposer des solutions de réutilisation, de recyclage ou toute autre forme de mise au rebut de cette unité dans un esprit de protection de l'environnement, lorsqu'il aura fini de l'utiliser.

# Instructions générales de sécurité

Les instructions suivantes servent de guide général d'installation et d'opération sécurisées des produits de télécommunications. Des instructions supplémentaires sont éventuellement indiquées dans le manuel.

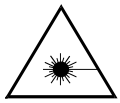
# Symboles de sécurité

***Avertissement***

Ce symbole peut apparaitre sur l'équipement ou dans le texte. Il indique des risques potentiels de sécurité pour l'opérateur ou le personnel de service, quant à l'opération du produit ou à sa maintenance.

Danger de choc électrique ! Evitez tout contact avec la surface marquée tant que le produit est sous tension ou connecté à des lignes externes de télécommunications.

Mise à la terre de protection : la cosse ou la borne marquée devrait être connectée à la prise de terre de protection du bâtiment.



***Avertissement***

Certains produits peuvent être équipés d'une diode laser. Dans de tels cas, une étiquette indiquant la classe laser ainsi que d'autres avertissements, le cas échéant, sera jointe près du transmetteur optique. Le symbole d'avertissement laser peut aussi être joint.

Veuillez observer les précautions suivantes :

- Avant la mise en marche de l'équipement, assurez-vous que le câble de fibre optique est intact et qu'il est connecté au transmetteur.
- Ne tentez pas d'ajuster le courant de la commande laser.
- N'utilisez pas des câbles ou connecteurs de fibre optique cassés ou sans terminaison et n'observez pas directement un rayon laser.
- L'usage de périphériques optiques avec l'équipement augmentera le risque pour les yeux.
- L'usage de contrôles, ajustages ou procédures autres que celles spécifiées ici pourrait résulter en une dangereuse exposition aux radiations.

ATTENTION : Le rayon laser peut être invisible !

Les utilisateurs pourront, dans certains cas, insérer leurs propres émetteurs-récepteurs Laser SFP dans le produit. Les utilisateurs sont avertis que RAD ne pourra pas être tenue responsable de tout dommage pouvant résulter de l'utilisation d'émetteurs-récepteurs non conformes. Plus particulièrement, les utilisateurs sont avertis de n'utiliser que des produits approuvés par l'agence et conformes à la réglementation locale de sécurité laser pour les produits laser de classe 1.

Respectez toujours les précautions standards de sécurité durant l'installation, l'opération et la maintenance de ce produit. Seul le personnel de service qualifié et autorisé devrait effectuer l'ajustage, la maintenance ou les réparations de ce produit. Aucune opération d'installation, d'ajustage, de maintenance ou de réparation ne devrait être effectuée par l'opérateur ou l'utilisateur.

# Manipuler des produits sous tension

## Règles générales de sécurité

Ne pas toucher ou altérer l'alimentation en courant lorsque le câble d'alimentation est branché. Des tensions de lignes peuvent être présentes dans certains produits, même lorsque le commutateur (s'il est installé) est en position OFF ou si le fusible est rompu. Pour les produits alimentés par CC, les niveaux de tension ne sont généralement pas dangereux mais des risques de courant peuvent toujours exister.

Avant de travailler sur un équipement connecté aux lignes de tension ou de télécommunications, retirez vos bijoux ou tout autre objet métallique pouvant venir en contact avec les pièces sous tension.

Sauf s'il en est autrement indiqué, tous les produits sont destinés à être mis à la terre durant l'usage normal. La mise à la terre est fournie par la connexion de la fiche principale à une prise murale équipée d'une borne protectrice de mise à la terre. Si une cosse de mise à la terre est fournie avec le produit, elle devrait être connectée à tout moment à une mise à la terre de protection par un conducteur de diamètre 18 AWG ou plus. L'équipement monté en châssis ne devrait être monté que sur des châssis et dans des armoires mises à la terre.

Branchez toujours la mise à la terre en premier et débranchez-la en dernier. Ne branchez pas des câbles de télécommunications à un équipement qui n'est pas mis à la terre. Assurez-vous que tous les autres câbles sont débranchés avant de déconnecter la mise à la terre.

## Connexion au courant du secteur

Assurez-vous que l'installation électrique est conforme à la réglementation locale.

Branchez toujours la fiche de secteur à une prise murale équipée d'une borne protectrice de mise à la terre.

La capacité maximale permissible en courant du circuit de distribution de la connexion alimentant le produit est de 16A. Le coupe-circuit dans l'installation du bâtiment devrait avoir une capacité élevée de rupture et devrait fonctionner sur courant de court-circuit dépassant 35A.

Branchez toujours le câble d'alimentation en premier à l'équipement puis à la prise murale. Si un commutateur est fourni avec l'équipement, fixez-le en position OFF. Si le câble d'alimentation ne peut pas être facilement débranché en cas d'urgence, assurez-vous qu'un coupe-circuit ou un disjoncteur d'urgence facilement accessible est installé dans l'installation du bâtiment.

Le disjoncteur devrait déconnecter simultanément les deux pôles si le système de distribution de courant est de type IT.

## Connexion d'alimentation CC

Sauf s'il en est autrement spécifié dans le manuel, l'entrée CC de l'équipement est flottante par rapport à la mise à la terre. Tout pôle doit être mis à la terre en externe.

A cause de la capacité de courant des systèmes à alimentation CC, des précautions devraient être prises lors de la connexion de l'alimentation CC pour éviter des courts-circuits et des risques d'incendie.

Les unités CC devraient être installées dans une zone à accès restreint, une zone où l'accès n'est autorisé qu'au personnel qualifié de service et de maintenance.

Assurez-vous que l'alimentation CC est isolée de toute source de courant CA (secteur) et que l'installation est conforme à la réglementation locale.

La capacité maximale permissible en courant du circuit de distribution de la connexion alimentant le produit est de 16A. Le coupe-circuit dans l'installation du bâtiment devrait avoir une capacité élevée de rupture et devrait fonctionner sur courant de court-circuit dépassant 35A.

Avant la connexion des câbles d'alimentation en courant CC, assurez-vous que le circuit CC n'est pas sous tension. Localisez le coupe-circuit dans le tableau desservant l'équipement et fixez-le en position OFF. Lors de la connexion de câbles d'alimentation CC, connectez d'abord le conducteur de mise à la terre à la borne correspondante, puis le pôle positif et en dernier, le pôle négatif. Remettez le coupe-circuit en position ON.

Un disjoncteur facilement accessible, adapté et approuvé devrait être intégré à l'installation du bâtiment.

Le disjoncteur devrait déconnecter simultanément les deux pôles si l'alimentation en courant CC est flottante.

# Quick Start Guide

If you are familiar with the VC-E1/16 and VC-E1/12 modules, use this guide to prepare a module for operation, and configure its parameters.

## 1. Preparing VC-E1/16, VC-E1/12 for Operation

### Selecting Module Operating Mode

Before installing the module, select its operating mode by means of switch SW2. See *Table 1* for a list of module operating modes.

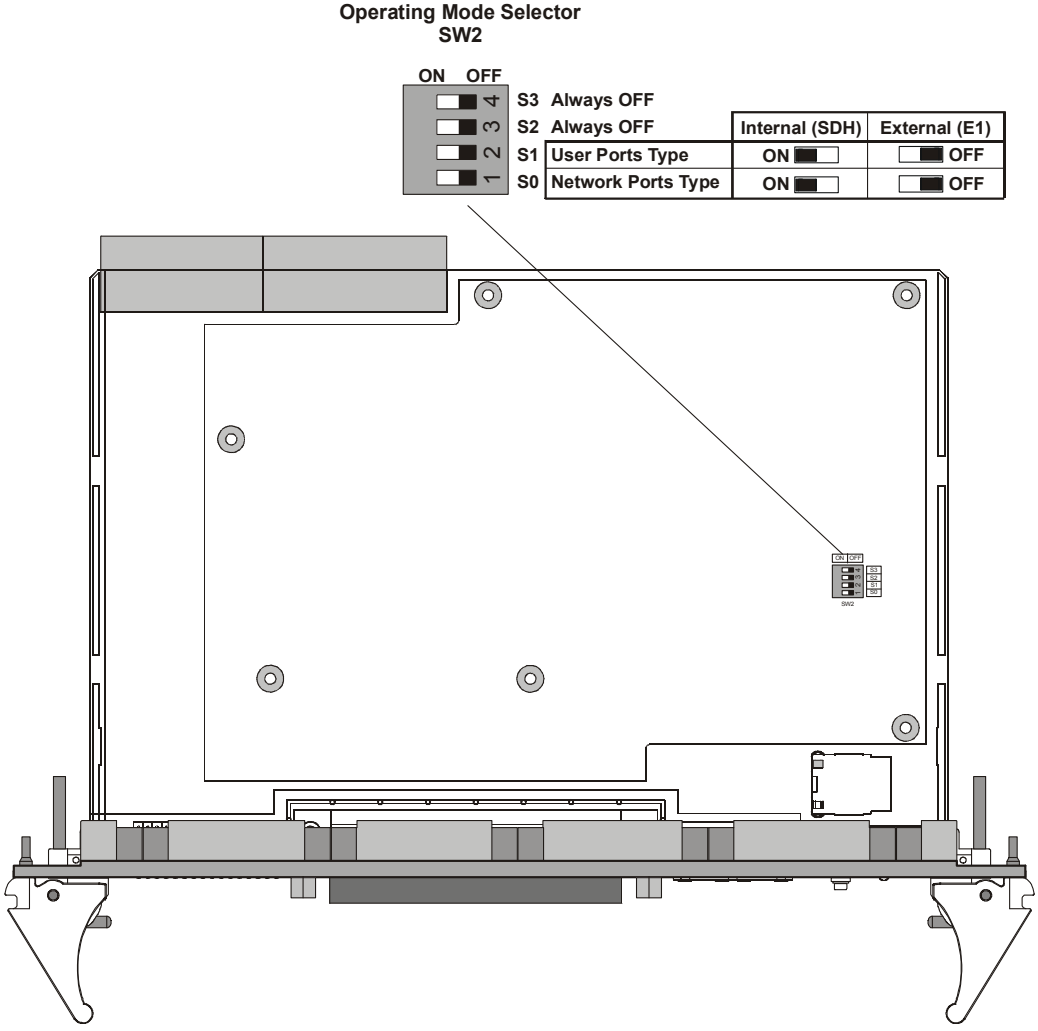
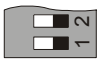
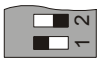
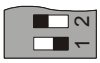
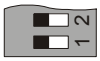


*Table 1. Operating Modes*

| Card Designation | User-Side (Voice) Ports | Network-Side (Data) Ports | Switch Settings |
|---|---|---|---|
| VC16E1_UE_NE, VC12E1_UE_NE | External ports (1-16E1 or 1-12 E1) | External data ports (LINK 1, LINK 2) |  |
| VC16E1_UE_NS, VC12E1_UE_NS | External ports (1-16E1 or 1-12 E1) | Internal (SDH) ports |  |
| VC16E1_US_NE, VC12E1_US_NE | Internal (SDH) ports | External data ports (LINK 1, LINK 2) |  |
| VC16E1_US_NS, VC12E1_US_NS | Internal (SDH) ports | Internal (SDH) ports |  |

## Installing the Module

Insert the module in the prescribed chassis slot: I/O-1 to I/O-5, I/O-7 or I/O-9.

## Connecting Cables

### Connecting Cables to External Voice Ports

➤ **To connect using the CBL-TELCO-RJ45/16E1 adapter cable (for equipment with balanced interfaces only):**

1. Connect the 64-pin connector of the CBL-TELCO-RJ45/16E1 adapter cable to the module voice port connector, 1-16 E1 or 1-12 E1.
2. Connect the grounding lug of the cable connector to the grounding screw on the module panel.
3. Route the cable through the cable guides provided in the rack to the required location.
4. Connect each RJ-45 plug (marked CH-1 to CH-16) to the prescribed user's equipment or patch panel connector. Insulate unused connectors, to prevent accidental short-circuiting of their exposed contacts to metallic surfaces.

➤ **To connect using an open cable (balanced or unbalanced interfaces):**

1. Route the open cable, CBL-TELCO-OPEN/2M or CBL-TELCO-OPEN/10M, to the required location, and then connect its free ends in accordance with the prescribed termination method.
2. Connect the 64-pin connector of the cable to the module voice port connector, 1-16 E1 or 1-12 E1.
3. Connect the grounding lug of the cable connector to the grounding screw on the module panel.

### Connecting Cables to External Data Ports

➤ **To connect equipment with balanced interfaces to an external data port:**

Use a standard RJ-45/RJ-45 cable.

➤ **To connect equipment with unbalanced interfaces to an external data port:**

1. Connect the RJ-45 male connector of the adapter cable, CBL-RJ45/2BNC/E1/X, to the prescribed module front panel LINK connector.
2. Connect coaxial cables to the prescribed user's equipment or patch panel connectors. Connect the other end of each cable to one of the adapter cable BNC plugs. Pay attention to correct connection:
   - The red connector is the transmit output of the link port
   - The green connector is the receive input of the link port.

# 2. Configuring VC-E1/16, VC-E1/12 Modules

## Configuring Module Operating Mode

If the module is not yet defined in the Gmux-2000 database, use the **Config>System>Card Type** screen to select the module type.

The selected mode must match the operating mode selected by means of switch SW2 (see *Table 1*).

Slots supporting VC-E1/16, VC-E1/12 modules: I/O-1 to I/O-5, I/O-7, and I/O-9.

## Configuring Module Global Parameters

### Configuring Signaling Profiles

1. Use the **Config>Physical Layer>I/O** screen of the module.
2. Select **Signaling Profile Config**.
3. Select **Profile Number to Configure**: 1 to 5.
4. Select translation rule in **ABCD BITS** row:
   - **A, B, C, D –** Bit value sent to local voice port is copied from the corresponding bit received from the network side.
   - **NOT A, NOT B, NOT C, NOT D –** Bit value sent to local voice port is inverted, relative to the value of the corresponding bit received from the network side.
   - **0** – Bit value is always 0.
   - **1** – Bit value is always 1.
5. Select the idle code: select bit values in the **IDLE SIGNAL** row.
6. Select the out-of-service code: select bit values in the **OOS SIGNAL** row.

### Bandwidth Management

➤ **To set the bandwidth control thresholds:**

1. On the **Config>Physical Layer>I/O** screen of the module, select **Bandwidth Control**.
2. Select **Q.50 Stop New Calls Threshold**, and then enter the global bandwidth utilization threshold at which a voice port starts rejecting new calls: the range is 25% to 98%.
3. Select **Discard Voice Packets Threshold** and then enter the global bandwidth utilization threshold at which a voice port starts discarding packets on active calls: the range is 25% to 99%.
4. Select **Block New Modem VBD Calls Threshold,** and then enter the global bandwidth utilization threshold at which a VC-E1/16, VC-E1/12 voice port starts blocking the set up of new voiceband modem calls: the range is 20 to 98%.
5. Select **Block New Modem Relay Calls Threshold,** and then enter the global bandwidth utilization threshold at which a VC-E1/16, VC-E1/12 voice port starts blocking the set up of new modem relay calls: the range is 20 to 98%.

➤ **To configure the keep-alive suppression rate:**

1. On the **Config>Physical Layer>I/O** screen of the module, select **SS7 Keep-Alive Suppression Rate**.
2. Select the fraction of FISUs and LSSUs to be suppressed: 0% (no suppression, is also suitable for protocols other than SS7) to 90%, in 10% increments.

## Configuring E1 Voice Port Physical Layer Parameters

Use **Config>Physical Layer>I/O>E1 Port Voice**.

| Parameter | Values | Parameter | Values |
|---|---|---|---|
| Channel ID | **1** to 16 or **1** to 12 | LIU Impedance | **BALANCE**<br>UNBALANCE |
| Admin Status | **CONNECTED**<br>NC | Profile | **1** through 5 |
| Idle Code | 00 to FF | Assign Entire Port to Bundle | 1 to 2000 |
| Restoration Time | **FAST**<br>CCITT<br>TR-62411 | Line Type | **FRAMED-MF**<br>FRAMED G.704<br>FRAMED G.704-CRC<br>FRAMED MF-CRC |
| Transmit CLK Source | **SYSTEM A**<br>LBT | Q.50 Protocol | **DISABLE**<br>ANNEX A<br>ANNEX B |
| Interface Type (for external ports only) | **LTU**<br>DSU | Q.50 Bits Pair | **A&B**<br>C&D |

# Configuring E1 Data Port Physical Layer Parameters

## Configuration Parameters for FRAMED Modes

Use **Config>Physical Layer>I/O>E1 Port Data**.

| Parameter | Values | Parameter | Values |
|---|---|---|---|
| Channel ID | **1** or 2 for external data ports<br>**1** to 4 for connection to SDH ports (with BACKUP STATUS = ENABLE) | Interface Type | **LTU**<br>DSU |
| Admin Status | **CONNECTED**<br>NC | Sub Channel Configuration | CHANNEL NUM:1 to 10<br>FUNCTION: DATA RELAY<br>CONNECT: YES |
| Backup Status | **DISABLE**<br>ENABLE | Assign Entire Time Slots to Port (for framed modes only) | Subchannel: 1 to 10 |
| Idle Code | 00 to **FF** | Line Type | **FRAMED G.704**<br>FRAMED G.704-CRC<br>UNFRAMED |
| Restoration Time | **FAST**<br>CCITT<br>TR-62411 | LIU Impedance (for external ports only) | **BALANCE**<br>UNBALANCE |
| Transmit CLK Source | **SYSTEM A**<br>LBT | | |

## Specific Configuration Parameters for UNFRAMED Mode

1. Select **Sub Channel Number: 1** to **10**.
2. Select **TS Type**:
   - **Connected**: normal operation, traffic can flow
   - **NC**: disables traffic flow through the subchannel, and thus through the whole data port.

## Configuring Backup for the Network-Side Connection

➤ **To configure the backup parameters:**

1. On the **E1 Port Data** screen, select **ENABLE** for **Backup Status**. The currently selected port becomes **Primary** port.
2. Select the **Backup Port**.
3. When necessary, repeat the procedure for the second pair of ports (applicable only for SDH ports).

# Configuring Parameters Related to SDH Transport

## Configuring VC-12 Path Parameters

Use **Config>Physical Layer>I/O>VC12 Channel**.

| Parameter | Values | Parameter | Values |
|---|---|---|---|
| Channel VC12 | VC16E1_US_NE: **1** to 16 for VC-E1/16,<br>**1** to 12 for VC-E1/12<br>VC16E1_UE_NS: **1** to 4<br>VC16E1_US_NS:<br>Voice: **1** to 16 for VC-E1/16,<br>**1** to 12 for VC-E1/12<br>Data: 17 to 20 | J2 Tx Path Trace | **DISABLE**<br>ENABLE |
| Connect State | **CONNECTED**<br>NC | J2 Rx Path Trace | **DISABLE**<br>ENABLE |
| AIS & RDI on Signal Label | **DISABLE**<br>ENABLE | J2 Path Trace | String of 15 alphanumeric characters (pad with spaces if necessary) |
| AIS & RDI on Path Trace | **DISABLE**<br>ENABLE | | |

## Mapping the Module Internal Ports to VC-12s

Use the **Config>Physical Layer>I/O>Telecom Assignment** screen for the prescribed STM1 module(s) to map each module internal port with **Connect State** set to **Connected**. For detailed instructions, refer to the Gmux-2000 Installation and Operation Manual.

# Configuring Bundle Parameters

## Preliminary Bundle Configuration Steps

Use **Config>Connection>Bundle**.

1. To modify an existing bundle, enter its number (1 to 2000), or select **Add** to define a new bundle.
2. Select **Config>Connection Mode**, and then select **TDMoIP (CV)**.
3. Select **PSN Type** to specify the type of packet-switched network:
   - **UDP/IP** – UDP over IP.
   - **MPLS/ETH** – MPLS over Ethernet.
4. Select **Config>Connection Configuration>Function** to specify packet structure:
   - For **UDP/IP:**
     - **TDMoIP+** – TDM over IP with RAD proprietary bandwidth-efficient packet structure
     - **TDMoIP** – TDM over IP with standard packet structure

- For **MPLS/ETH:**
  - **AAL2oMPLS+** – AAL2 over MPLS with RAD proprietary bandwidth-efficient packet structure
  - **AAL2oMPLS** – AAL2 over MPLS with standard packet structure

5. Select the **Connection State** of the bundle: **Enable** or **Disable**.

## Configuring Bundle Routing Parameters

Use **Config**›**Connection**›**Connection Configuration**›**Routing Parameters**.

| Parameter | Values | Parameter | Values |
|---|---|---|---|
| Destination IP | Valid IP in dotted-quad notation | Exit Port | Slot/Port, where:<br>VC-E1/16 data port:<br>Slot: 1 to 5, 7, 9<br>Port: 17 or 18<br>GbE module port:<br>Slot: 6 or 8<br>Port: 1 or 2 |
| Next Hop | Valid IP in dotted quad notation | Exit Channel | **1** to 10 |
| Source CBID | UDP/IP: 1 to 8063<br>MPLS/ETH: 1 to 8063 | TOS | **00** to FF |
| Dest CBID | UDP/IP: 1 to 8063<br>MPLS/ETH: 1 to 8063 | VLAN Parameters | VLAN Tagging: **DISABLE**, ENABLE<br>For ENABLE:<br>VLAN ID: 1 to 4094<br>VLAN Priority: 7 to **0** |

## Configuring General Bundle Parameters

1. Configure bundle packetizing parameters:

| Parameter | Values |
|---|---|
| Max Bytes in Multiplexed Frame | 100 to 1461 |
| Transparent Jitter Size | N*10 msec,<br>where N = **2** to 10 (**20** to 100 msec) |
| Packetizing Interval | 10 to 90 |

2. Configure **Connectivity Parameters**:

| Parameter | Values |
|---|---|
| Connectivity Packet Rate (Sec) | 1 to **60** |
| Connectivity Timeout Cycles | 2 to 5 |

## Configuring Bundle Voice Parameters

Use **Config>Connection>Connection Configuration>Voice Parameters**.

| Parameter | Values | Parameter | Values |
|---|---|---|---|
| Coder/Rate (kbps) | **G.723.1/6.4**<br>G.723.1/5.3<br>G.729A/8<br>Only for Super Tandem = DISABLE<br>G.711 (A-law)<br>G.711 (u-law) | Custom Tone Detection | See below |
| Caller ID Type | **Bellcore Type 1**<br>V.23 | Super Tandem | **DISABLE**<br>ENABLE |
| Tx Delay for Caller ID | 0 to 3000 | Echo Canceller | **ENABLE**<br>DISABLE |
| Volume to Line | -6 dB to +6 dB in 1-dB steps | Coding | **A-Law**<br>**U-Law** |
| Volume from Line | -12 dB, -6 to +6 dB in 1-dB steps, +12 dB | VAD | See below |
| Fax/Modem | See below | CNG Mode | **ENABLE**<br>DISABLE |
| MF Parameters | MFCR2 RELAY DISABLE<br>MFCR2 RELAY ENABLE<br>See below | Advanced Voice Parameters | See below |

## Fax/Modem

| Parameter | Values | Parameter | Values |
|---|---|---|---|
| Modem | **DISABLE**<br>ENABLE | Fax Rate (kbps) | **14.4 kbps**<br>4.8 kbps<br>9.6 kbps<br>VBD |
| Modem Op Mode | **VOICEBAND DATA**<br>RELAY | Max. VBD Modem Calls | 1 to **62** |
| VBD Rate | **G.711 64k 5 msec**<br>G.711 64k 10 msec<br>G.726 32k<br>G.726 24k | Max. Relay Modem Calls | 1 to **62** |
| Fax | **ENABLE**<br>DISABLE | | |

## MF Parameters

| Parameter | Values |
|---|---|
| Minimum Pulse Width (msec) | 45 to 300 |
| Minimum Power Level (dBm) | -1 to -35 |
| MFC Spoofing | **DISABLE**<br>ENABLE |
| Tone ACK Interval | N*20 msec,<br>where N = **3** to 20 (**60** to 400 msec) |

## Custom Tone Detection

| Parameter | Values |
|---|---|
| **Tone Detection** | **DISABLE**<br>ENABLE |
| **Tone Frequency** | For ENABLE:<br>**2000 Hz**<br>1780 Hz<br>2000 Hz+1780 Hz |

## VAD

| Parameter | Values |
|---|---|
| VAD Method | **Generic VAD Policy**<br>Internal VAD Policy |
| Noise Level for VAD | For Generic VAD Policy:<br>**LOW**<br>OFF<br>HIGH |

## Advanced Voice Parameters

| Parameter | Values |
|---|---|
| Cdis/CNG Detection Time | 10 to 100 msec (in multiples of 10 msec) |

## Configuring Timeslot Assignment for Voice Ports

Use **Config>Connection>TS Assignment**.

➤ **To assign individual voice port timeslots:**

1. Select existing **Bundle**: range is **1** to **2000**.
2. To assign individual timeslots, select **Ts Type**:

| | |
|---|---|
| **NC** | Timeslot is not connected to the selected bundle. |
| **VOICE** | Voice timeslot (timeslot with CAS information). |
| **HDLC-1** | Timeslot whose payload is interpreted as HDLC data (timeslot that carries CCS data). |
| **SS7-1** | Timeslot whose payload is interpreted as being signaling data per Signaling Scheme 7. |
| **Trans** | Transparent timeslot, that is, a timeslot whose payload is transferred without any processing to the far end. |

3. For connections ending at a Vmux Voice Trunking Gateway port, for each timeslot specify the destination port and timeslot:
   - Select the destination port of the timeslot at the Vmux side using **Dest Port**
   - Select the destination timeslot at the Vmux side using **Dest Ts**.

➤ **To assign a range of timeslots to a bundle:**

1. Select **Assign TS Range to Bundle**.
2. Select existing **Bundle**: range is **1** to **2000**.
3. Select **TS Type** and specify the type assigned to all the timeslots in the group.
4. Select **Dest Port** and enter the prescribed far end port number.
5. Select **From Ts** and specify the first timeslot in the range to be assigned.
6. Select **Num of TS** and enter the number of timeslots to be connected to the selected bundle.
7. Confirm the connection by selecting **Connect the TS Range to Bundle.**

➤ **To assign all the voice port timeslots to one bundle:**

1. Select **Assign Entire Port to Bundle**.
2. Select existing **Bundle**: range is **1** to **2000**.
3. Select **Ts Type** and specify the type assigned to all the port timeslots.
4. Select **Dest Port** and enter the prescribed far end port number.
5. Confirm by selecting **Connect Entire Port to Bundle**.

## Configuring Timeslot Assignment for Data Ports

Use **Config>Connection>TS Assignment**.

➤ **To assign individual data port timeslots:**

1. Select **Sub Channel**: range is **1** to **10**.
2. To assign individual timeslots, select **Ts Type**: **CONNECTED** or **NC** (not connected).

➤ **To assign all the data port timeslots to one bundle:**

1. Select **Assign Entire TSs to Port**.
2. Select **Sub Channel**: range is **1** to **10**.
3. Confirm by selecting **Connect Entire Time Slots to Port**.

## Configuring N+1 Protection for VC-E1/16 Modules

1. Open the **Config>System>Redundancy>I/O** screen and type **X** to add a new APS group.
2. Open the **APS Configuration** screen for the new APS group, and select **N+1** for **Config Mode**.
3. Select the desired **Recovery Mode**: **Revertive** or **Non-Revertive**. For the **Revertive** mode, specify the **Time to Restore** value.
4. Open the **APS Mapping** screen, and enter the slot of the protection VC-E1/16 or VC-E1/12 module in the **CH NUM 1** row (leave the port field value as **0**).
5. Enter the slots of the protected VC-E1/16, respectively VC-E1/12, modules in the other rows.
6. Select for each module its protection **Priority**: **Low** or **High**.

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# Chapter 1
# Introduction

## 1.1 Overview

### Purpose

This manual describes the technical characteristics, applications, installation and operation of the VC-E1/16 and VC-E1/12 voice compression modules for the Gmux-2000 Pseudowire Gateway.

VC-E1/16 and VC-E1/12 modules enable Gmux-2000 to transport compressed E1 voice traffic, including inband signaling, modems and fax, over TDM links (E1 and STM-1) and Gigabit Ethernet packet-switched networks using IP and/or MPLS transport. Advanced systems design confers the flexibility needed to meet the requirements of practically every voice transport application.

The modules are fully compatible with RAD Vmux family of Voice Trunking Gateways, and can be managed by the RADview-SC/Vmux Service Center for Vmux Applications (version 4.5 or higher), the same service center used to manage the Vmux product family.

### Models

The voice compression modules are available in two models, which differ only in the number of voice ports:

- VC-E1/16 – module with 16 E1 voice ports
- VC-E1/12 – module with 12 E1 voice ports.

***Note***

*Unless specifically stated otherwise, the generic term **VC-E1/16** is used when the information is applicable to both module models. Information applicable to a specific model is explicitly indicated.*

### Main Features

VC-E1/16 and VC-E1/12 are complete voice compression subsystems with a compression capacity of 16 E1, respectively 12 E1, voice trunks.

The modules enable Gmux-2000 systems to deliver voice traffic, including inband signaling (DTMF, MFR2, and MFC), voiceband modems and Group III fax up to 14.4 kbps, over the following types of links:

- TDM links: E1 (PDH) links located on the VC-E1/16 module itself, or STM-1 (SDH) ports located on STM1 modules

- GbE links, located on GbE modules.

In addition to voice traffic, VC-E1/16 can also transparently transfer selected timeslots.

VC-E1/16 supports the ITU-T Rec. G.723.1 and G.729 Annex A voice compression algorithms, as well as ITU-T Rec. G.711 (A-law and μ-law PCM), and uses TDMoIP multiplexing for efficiently transporting the payload of E1 trunks over TDM (E1 or SDH) networks. Packet structure is selectable (TDMoIP for UDP/IP networks, and AAL2oMPLS for MPLS networks). In addition to standard packet structures, VC-E1/16 also offers RAD proprietary versions of these protocols that achieve significantly higher bandwidth efficiency.

Advanced digital signal processing techniques ensure highly-reliable service, unaffected by the use of voice compression. These techniques include:

- Per-channel voice activity detection (VAD) and silence suppression
- Built-in echo cancellation for delays up to 32 msec per ITU-T Rec. G.168
- Relaying of Group III fax, modems at commonly used rates and standards, inband signaling and custom tones.
- Where compression cannot be used, for example, for non-standard inband signaling and modem protocols, voiceband signals can be faithfully transmitted using selectable PCM or ADPCM coders.

To prevent voice degradation when traffic passes multiple VC-E1/16 or Vmux units, a special super-tandem mode ensures that voice signals pass compression only once, irrespective of the number of hops on the transmission path.

At the highest compression ratio, VC-E1/16 enables compressing up to 16 E1 voice trunks for transport over a single E1 network link (up to 16:1 compression ratio); for VC-E1/12, the maximum compression ratio cannot exceed 12:1 because it has only 12 voice ports. VC-E1/16 and VC-E1/12 traffic can be routed over up to 32 independently-configurable bundles, where a bundle can carry any number of timeslots, up to a full E1 voice port (30 or 31 channels). Each compressed voice bundle can be independently connected to any desired network-side port (either TDM port or GbE port).

Moreover, VC-E1/16 and VC-E1/12 support both point-to-point and point-to-multipoint (channelized) applications over their E1 network-side links, with user-configurable link bandwidth assignment per destination (the timeslots available on each E1 network-side link can be assigned to up to 10 independent subchannels).

To guarantee high service quality together with highly efficient utilization of transport links, VC-E1/16 and VC-E1/12 support the ITU-T Rec. Q.50 (Annex A and B) bandwidth control protocol, as well as a proprietary bandwidth limiting algorithm that ensures quality degrades gradually during intervals of excessive traffic load. To preserve service quality, a separate mechanism makes it possible to block the setup of new modem calls when bandwidth utilization exceeds user-configurable, application-specific thresholds.

VC-E1/16 and VC-E1/12 provide full support for a wide range of signaling protocols. This includes CAS and R2 signaling (transferred end-to-end either transparently, or after translation in accordance with user-specified profiles) as well as optimized HDLC-based transfer of CCS protocols such as ISDN, QSIG,

Signaling Scheme 7 (SS7), etc. The CCS timeslots are user-selectable, independently for each voice port, with a maximum of two timeslots per port.

VC-E1/16 and VC-E1/12 have both TDM and packet ports:

- The TDM ports comprise a set of 16, respectively 12, voice (user-side) E1 ports and a separate set of 2 data (network-side) E1 ports
- The packet ports comprise 2 network-side ports, for connection to Gmux-2000 internal packet buses.

To provide maximum application flexibility and avoid the need for additional equipment, each set of TDM ports has both external ports with ITU-T Rec. G.703 interfaces located on the module panel, and internal ports that can be connected via the Gmux-2000 telecom buses to the SDH ports located on the STM1 modules that may be installed in the chassis (each internal module port can be independently mapped to any desired VC-12).

The maximum number of VC-E1/16 and VC-E1/12 modules that may be installed in a Gmux-2000 chassis is 7. Note however that 2 out of the 7 Gmux-2000 I/O slots that support VC-E1/16 modules also support STM1 modules, and therefore the maximum voice compression capacity (7 modules) can be reached only when using the external VC-E1/16 or VC-E1/12 module ports (when using internal ports, STM1 modules must be installed in the chassis, thereby reducing the number of slots available for modules by one for each STM1 module installed). The maximum number of VC-E1/16 or VC-E1/12 modules decreases by one for each installed STM1 module:

- With VC-E1/16 modules, the maximum capacities are as follows:
  - Without STM1 modules in the chassis, 7 VC-E1/16 modules yield 112 E1 voice trunks per chassis. Assuming that one timeslot is reserved for signaling (for example, timeslot 16 for CAS, or any timeslot for CCS), the number of voice timeslots per E1 trunk is 30, resulting in a maximum capacity of 3360 voice channels per module.

    When CCS protocols such as SS7 are used, the signaling information can be separately carried, in accordance with customer's signaling transfer method: in this case, the number of voice timeslots per trunk can reach 31, for a maximum capacity of 3472 channels per module.
  - One STM1 module and 6 VC-E1/16 modules yield a maximum capacity of 96 E1 trunks per chassis (2880, respectively 2976, voice channels)
  - Two STM1 modules and 5 VC-E1/16 modules yield a maximum capacity of 80 E1 trunks per chassis (2400, respectively 2480, voice channels).
- With VC-E1/12 modules, the respective maximum capacities are 84, 72, and 60 E1 voice trunks (up to 2520, 2160, and 1800 voice channels for 30 timeslots per trunk, or 2604, 2232, and 1860 for 31 timeslots per trunk).

To improve availability, VC-E1/16 and VC-E1/12 data ports can be configured to provide backup for the connection to the network. In this case, the data ports operate in pairs: in each pair, one port serves as the main (primary) port, and the other as backup (secondary) port. At any time, only one port of each pair carries traffic. As long as the primary port can carry the traffic, it is automatically selected as the active port; in case of a problem on the link or in the port

hardware, the traffic is automatically switched to the backup port. Switching is revertive.

This means that when using backup over the external E1 data (network-side) ports, only one E1 port is available; when using the internal (SDH) network ports with backup enabled, an additional set of internal data ports becomes available, and therefore the maximum number of E1 data ports is four (two primary and two secondary).

For maximum service availability, Gmux-2000 offers N+1 protection for I/O modules. This protection mode, described in detail in Gmux-2000 Installation and Operation Manual, is available for VC-E1/16 modules configured to use only internal ports for both voice and data. In the N+1 protection mode, one VC-E1/16 module is assigned to serve as a hot standby for the other VC-E1/16 modules installed in the chassis.

VC-E1/16 operating parameters are determined by commands received from the Gmux-2000 CONTROL module. The CONTROL module can also download new software to the module, when the Gmux-2000 software is updated. VC-E1/16 modules also support inband management of remote Gmux-2000 units: a remotely located network management station, for example, RADview-SC/Vmux, connected to a GbE port of one Gmux-2000 unit can manage the other Gmux-2000 units interconnected to it via VC-E1/16 data ports. Vmux-2100 Voice Trunking Gateways connected to VC-E1/16 ports can also be managed in the same way.

The module supports comprehensive diagnostics, including power-up self-test, local and remote loopbacks and test tone injection for the voice ports, and remote loopbacks for the network ports. It also collects comprehensive performance statistics that can be uploaded via TFTP to a management station such as the RADview-SC/Vmux for analysis. Front-panel indicators indicate at a glance the status of each module port.

## 1.2 Physical Description

Each VC-E1/16 or VC-E1/12 module occupies a single I/O slot in the Gmux-2000 chassis. The modules can be installed in slots IO-1 to IO-5, IO-7 and IO-9.

The only internal setting is the selection of the module operating mode; all the module configuration parameters are determined by software.

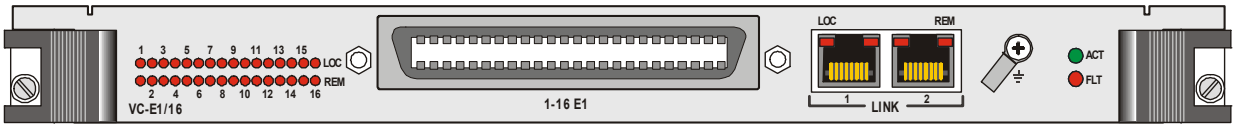
### Module Panels

*Figure 1-1* shows typical VC-E1/16 and VC-E1/12 module panels.

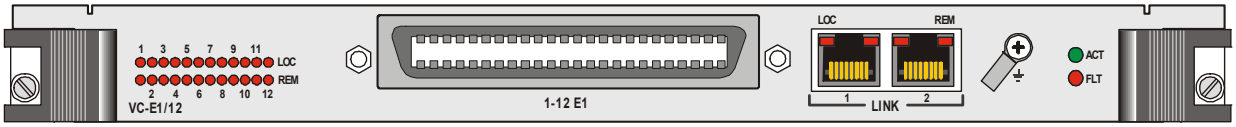
The panel includes the following main components:

- 64-pin TELCO female connectors, designated 1-16 E1, respectively 1-12 E1, for the external voice ports
- 16, respectively 12, pairs of alarm indicators, one pair for each voice port
- Two RJ-45 connectors with built-in alarm indicators, designated LINK 1 and LINK 2, for the external data ports

- Grounding screw, for connecting the shield of the multipair cable that carries the voice ports
- Two module status indicators.




*A. VC-E1/16 Panel*




*B. VC-E1/12 Panel*

*Figure 1-1. Typical Module Panels*

## Indicator Functions

### Port Alarm Indicators

Each module port (either voice or data) has two alarm indicators:

- **LOC –** Lights when the input signal level is too low, or the port loses frame synchronization (including loss of synchronization in case AIS is received)
- **REM –** Remote loss of signal indicator, lights when the remote E1 port reports loss of synchronization.

### Module Status Indicators

The functions of the module status indicators are as follows:

**ACT**
- Flashes during module initialization and during software downloading.
- Lights steadily after initialization is successfully completed, if no fault is detected in the module.

**FLT** Lights when a malfunction has been detected in the module.

## 1.3 Typical Applications

This section provides typical applications for VC-E1/16 modules. For additional applications, or an analysis of your particular application, contact RAD Marketing Department.

# High Capacity Toll Bypass

### Toll Bypass over SDH Networks

*Figure 1-2* shows a Gmux-2000 equipped with VC-E1/16 modules used to provide a high-capacity toll bypass link over the SDH network.

In this application, a high-capacity PBX, located on customer's premises, may be directly connected to a PSTN switch via a large number of E1 trunks. The maximum number of E1 trunks depends on the desired protection level for the STM-1 link:

- When only one STM1 module is used (line redundancy only), the maximum number of E1 trunks supported by one Gmux-2000 chassis is 96 (up to 2880 or 2976 compressed voice channels, depending on the number of voice timeslots per trunk, 30 or 31)
- When two STM1 modules are used (line and hardware redundancy), the maximum number of E1 trunks supported by one Gmux-2000 chassis is 80 (up to 2400 or 2480 compressed voice channels, depending on the number of voice timeslots per trunk, 30 or 31).

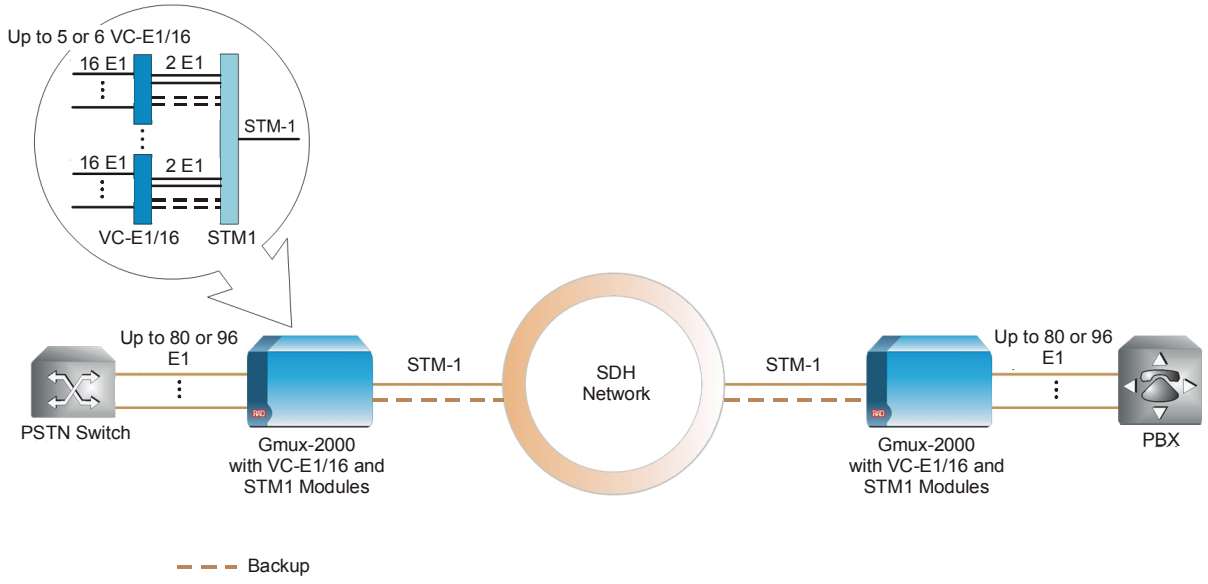



*Figure 1-2. High Capacity Toll Bypass over SDH Network*

### Toll Bypass over Packet Switched Networks

*Figure 1-3* shows a Gmux-2000 equipped with VC-E1/16 modules used to provide a high-capacity toll bypass link over a packet switched network.

This application uses the packet ports of the VC-E1/16 modules to direct the compressed voice traffic (or any part of this traffic, as configured by the user) to the desired GbE port(s) of GbE modules installed in the Gmux-2000. The PSN connection characteristics are configured by defining bundles.

VC-E1/16 can operate over IP and/or MPLS packet switched networks, in accordance with user's configuration for each bundle.

PSN connections can be used in parallel with any type of TDM network connections.

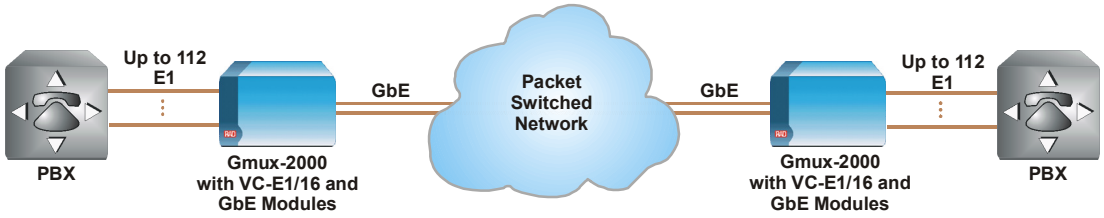



*Figure 1-3. High Capacity Toll Bypass over Packet Switched Network*

## Connecting Multiple Remote Call Centers

*Figure 1-4* shows a Gmux-2000 equipped with VC-E1/16 modules that provides access links to from the corporate PBX or from a PSTN switch to several remote call centers, each using one Vmux-2100 Voice Trunking Gateway.

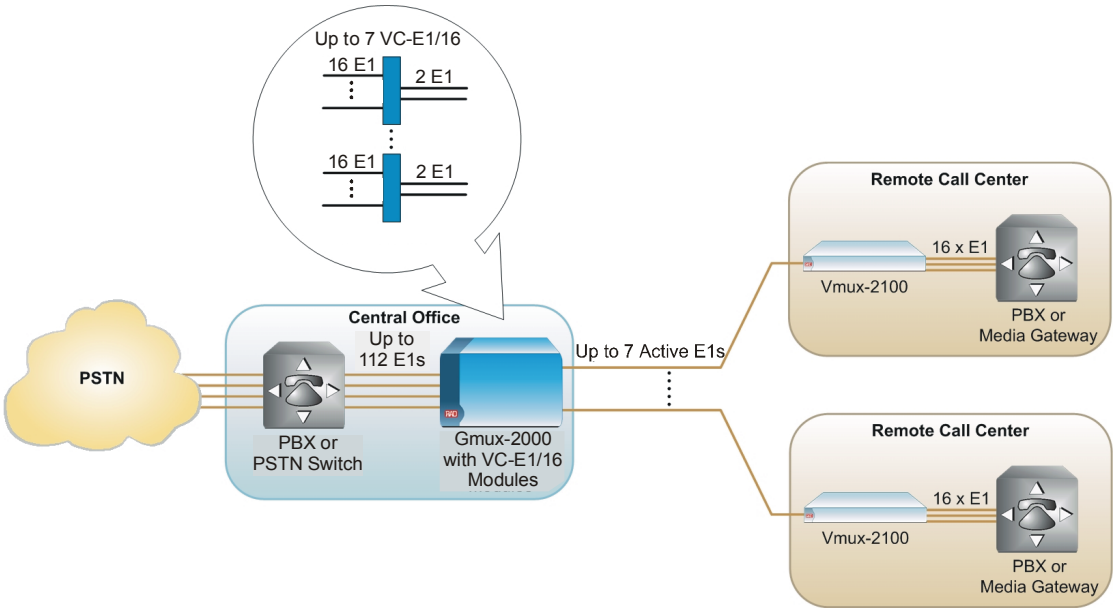



*Figure 1-4. Remote Call Centers (Main Link Hot Standby Mode)*

When equipped with VC-E1/16 modules, the maximum capacity of the Gmux-2000 is 112 E1 trunks, using only one E1 link to connect a group of up to 16 E1 trunks to each call center.

The connection to the Vmux-2100 units is made over the external data ports. In this application, a separate VC-E1/16 module is used for each remote location: in this way, availability can be greatly enhanced by using the backup function: each E1 link can be backed up by using the other data port.

Using the point-to-multipoint capability of VC-E1/16 modules enables connecting to an even larger number of remote locations, because each VC-E1/16 data port can connect to up to 10 separate locations.

VC-E1/16 also supports inband management of the Vmux-2100 units: in *Figure 1-4*, a network management station, for example, RADview-SC/Vmux, connected to a GbE port of the Gmux-2000 unit can manage inband the Vmux-2100 units connected via VC-E1/16 data ports.

## Voice Trunking over Multiple E1 Streams

*Figure 1-5* shows a Gmux-2000 equipped with VC-E1/16 modules that provides voice trunks between two MSCs with STM-1 interfaces over E1 infrastructure, either as a replacement for access to SDH infrastructure, or when no SDH infrastructure is available.

The maximum number of E1 trunks depends on the desired protection level for the STM-1 link to each MSC:

- When only one STM1 module is used (line redundancy only), the maximum number of E1 trunks supported by one Gmux-2000 chassis is 96
- When two STM1 modules are used (line and hardware redundancy), the maximum number of E1 trunks supported by one Gmux-2000 chassis is 80.

VC-E1/16 also supports inband management of the Gmux-2000 units: in *Figure 1-5*, a network management station, for example, RADview-SC/Vmux, connected to a GbE port of one Gmux-2000 unit can manage inband the other Gmux-2000 unit, via the VC-E1/16 data ports.

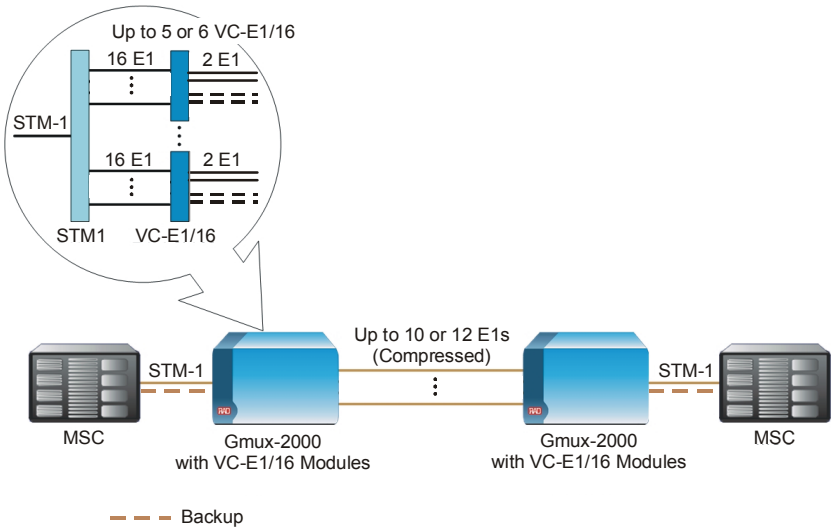



*Figure 1-5. Voice Trunking over Multiple E1 Streams (Main Link Redundancy Mode)*

## Voice Trunking over STM-1 Link

*Figure 1-6* shows a Gmux-2000 equipped with VC-E1/16 modules that provides voice trunks from a remote location over an STM-1 link.

Using voice compression reduces the number of VC-12s needed on the STM-1 link, thereby freeing bandwidth for other applications.

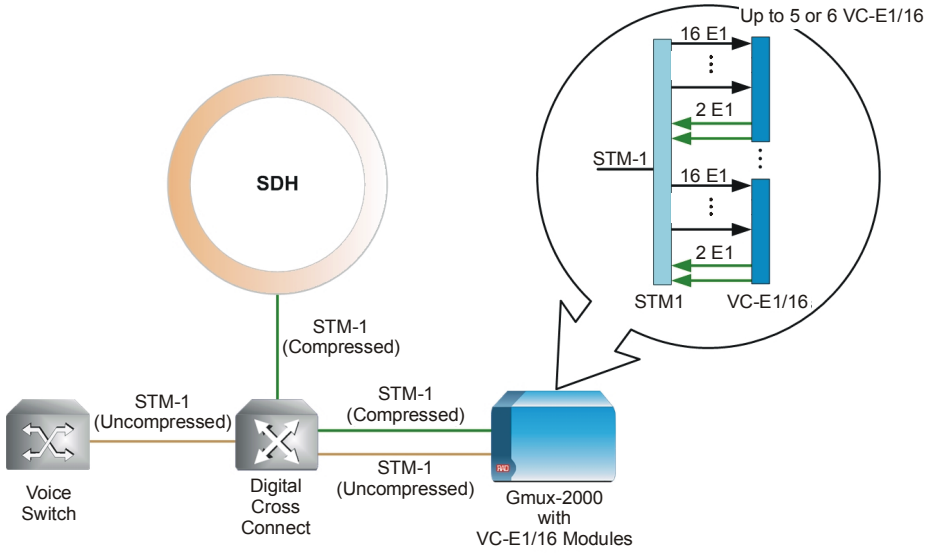



*Figure 1-6. Voice Trunking over STM-1 Link*

## Interoperability with Vmux Family Products

VC-E1/16 modules are fully interoperable with the RAD Vmux family of Voice Trunking Gateways. *Figure 1-7* shows a Gmux-2000 equipped with VC-E1/16 modules connecting a large number of E1 voice trunks to multiple locations, using a much smaller number of E1 links to carry the compressed voice traffic.

An important advantage of VC-E1/16 is that it provides for inband management of the individual Vmux units by means of the same network management station, for example, RADview-SC/Vmux, that is used to manage the Gmux-2000 in which the VC-E1/16 module is installed.

The management station can connect to the Gmux-2000 from a remote location, via the GbE module.

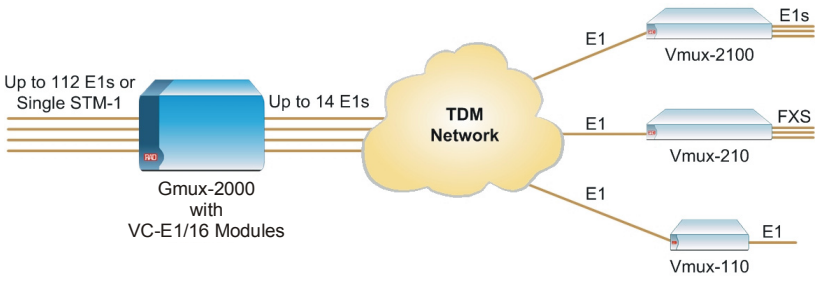



*Figure 1-7. Interoperability with Vmux Equipment over TDM Networks*

*Figure 1-8* shows a similar application that uses an IP network for connecting to the Vmux-2100 units.

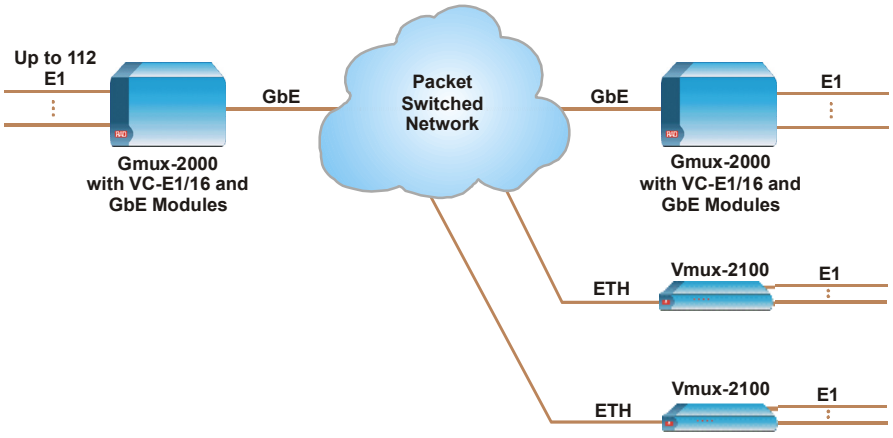



*Figure 1-8. Interoperability with Vmux Equipment over PSN Networks*

## 1.4 Functional Description

### Functional Block Diagram

*Figure 1-9* shows the functional block diagram of the VC-E1/16 module. The functional block diagram of the VC-E1/12 module is similar, except that it has only 12 E1 voice ports. See description of operating modes in the *VC-E1/16 Operating Modes* section on page *1-16*.

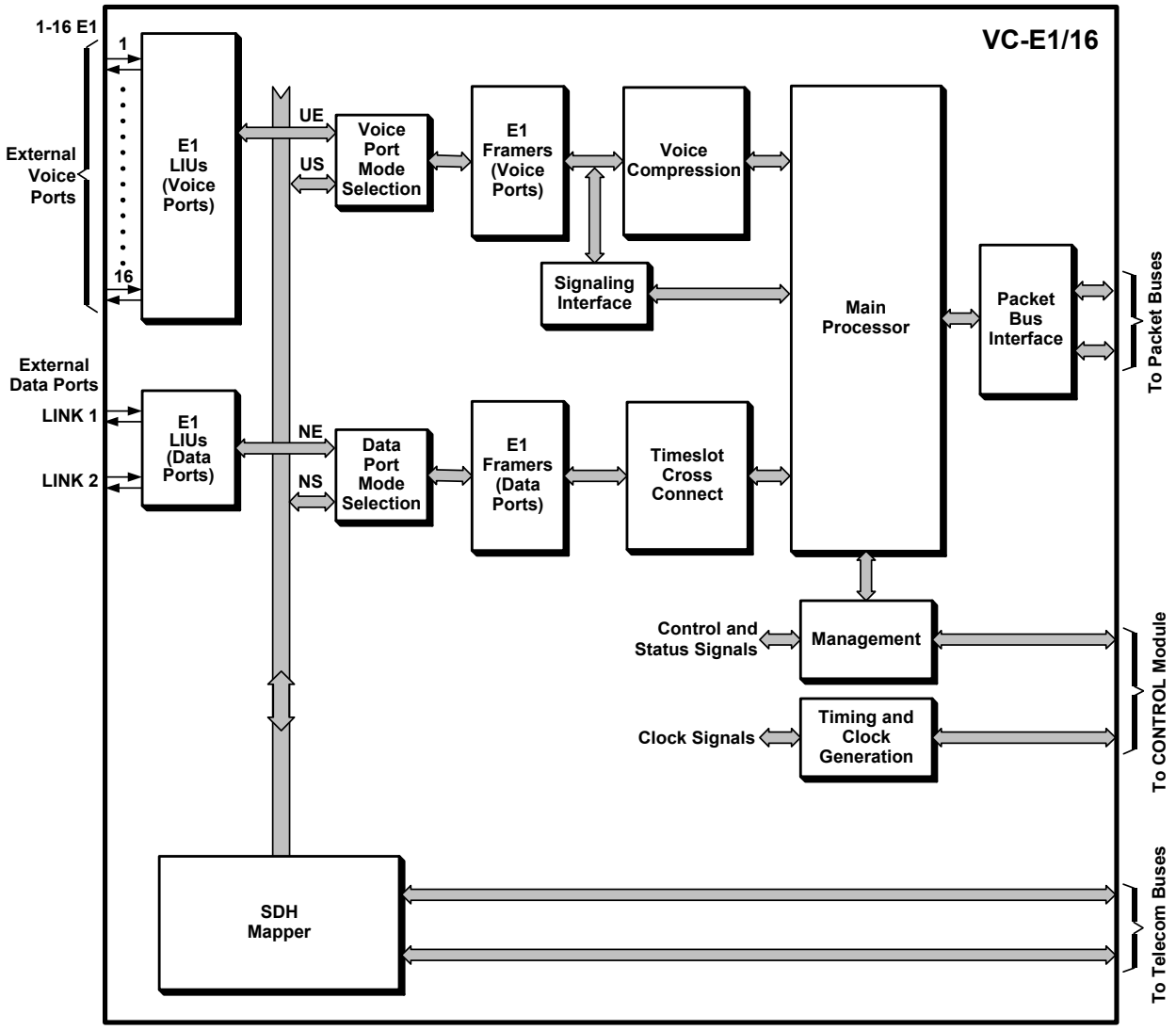



*Figure 1-9. VC-E1/16 Module, Functional Block Diagram*

The operation of the main VC-E1/16 subsystems is described below.

## User Interfacing Subsystem

The user interfacing subsystem includes 16 voice ports, each comprising an E1 framer and a port interface.

- In the transmit path, each voice port synchronizes to the incoming E1 stream, terminates timeslot 0, collects the payload from the incoming E1 timeslots, and sends the audio signals to the voice compression subsystem. The port signaling information is collected by the signaling interface and sent to the main processor for processing.
- In the receive path, each port collects the decompressed audio signals from the voice compression subsystem and the restored signaling information from the signaling interface, adds the framing overhead, and rebuilds the E1 frame in accordance with the selected framing mode.

The voice ports support the following framing modes:

- Basic G.704 framing, with or without support for CRC-4 in accordance with ITU-T Rec. G.704 and G.706. The CRC-4 option should be selected whenever possible, because it allows to monitor the E1 link transmission performance.

  The basic G.704 framing mode is required for applications that require CCS. The user can specify the timeslots that carry the CCS protocol (up to two timeslots). These timeslots are then transferred to the main processor, for processing. The supported CCS protocols are ISDN, QSIG and SS7.

- G.704 framing with timeslot 16 multiframe (G.704 multiframe mode), with or without support for CRC-4. This framing mode is required for applications that require CAS.

The voice ports have two sets of interfaces:

- External physical interfaces terminated in the 1-16E1 or 1-12E1 connector, for directly connecting to the user's PBX/voice switch.

  Each voice port has an ITU-T Rec. G.703 interface with independently-selectable characteristics:

  - 120 Ω balanced line interface.
  - 75 Ω unbalanced interface.

  For each port, only one of these two interfaces is active at any time.

  The maximum allowed line attenuation is 36 dB for the LTU mode, and 12 dB for the DSU mode.

- Internal interfaces (logical ports) that connect through the VC-E1/16 SDH mapper to the Gmux-2000 telecom buses. These logical ports are mapped to VC-12s on the desired SDH ports, and the path parameters can be configured as for any other VC-12 ports.

At any time, only one set of voice port interfaces (either the external or the internal set) is active, that is, connected to the E1 voice framers, in accordance with the selected operating mode (see *VC-E1/16 Operating Modes* section on page *1-16*).

## Network Interfacing Subsystem

The network interfacing subsystem consists of data ports, each comprising an E1 framer and a port interface.

- In the transmit path, each data port collects the compressed audio signals from the main processor and builds the data stream for transmission to the network.
- In the receive path, each data port synchronizes to the incoming data stream, collects the payload and sends it to the main processor, for further processing (separation of signaling and compressed audio).

The main processor provides the information to be sent to the network, and it also handles the information received from the network. The connection between the data ports and the main processor is made by a timeslot cross-connect matrix.

The data ports support the following framing modes:

- Basic G.704 framing mode, with or without support for CRC-4 in accordance with ITU-T Rec. G.706, with timeslot 0 terminated at each port.

  This mode provides 31 payload timeslots per port (total bandwidth available for payload – 1984 kbps), and has the advantage that it supports channelizing for point-to-multipoint applications. The number of subchannels that may be defined on each port is up to 10 (the minimum is 1). Each subchannel operates essentially as a transparent pipe for a preconfigured fraction of the total payload.

- Unframed mode. This mode provides 32 payload timeslots per port, and therefore enables full utilization of link bandwidth (2048 kbps), but can be used only for point-to-point applications (no channelizing – the whole port is handled as a single subchannel).

The data ports have two sets of interfaces:

- External physical interfaces terminated in the LINK connectors, for connecting to E1 transmission equipment. The number of external E1 data ports is 2.

  Each data port has an ITU-T Rec. G.703 interface with independently-selectable characteristics:

  - 120 Ω balanced line interface.
  - 75 Ω unbalanced interface.

  For each data port, only one of these two interfaces is active at any time.

  The maximum allowed line attenuation is 36 dB for the LTU mode, and 12 dB for the DSU mode.

- Internal interfaces (logical ports) that connect through the VC-E1/16 SDH mapper to the Gmux-2000 telecom buses. The number of internal data ports is either 2 (without backup) or 4 (backup enabled). These logical ports are mapped to VC-12s on the desired SDH ports, and the path parameters can be configured as for any other VC-12 ports.

At any time, only one set of data port interfaces (either the internal or the external ports) is active, that is, connected to the E1 data framers, in accordance with the selected operating mode (see *VC-E1/16 Operating Modes* section on page *1-16*).

To improve service availability, the network interfacing subsystem can be configured to provide backup for the connection to the network. In this case, the data ports operate in pairs. This means that when using the external E1 data ports, the maximum bandwidth of the network connection is the bandwidth available on a single port (1984 kbps for framed modes, 2048 kbps for the unframed mode). When connecting through the internal (SDH) ports, the maximum bandwidth is the bandwidth available on two ports.

## Packet Bus Interfacing Subsystem

The packet bus interfacing subsystem includes an Ethernet switch that handles the traffic flow between the internal packet ports of the main processor and the Gmux-2000 packet buses, en route to the GbE PSN interface module. This traffic flow is configured by defining bundles terminated on GbE ports.

The internal packet ports are connected to the chassis packet buses via two Fast Ethernet transceivers.

## Support for N+1 Protection

VC-E1/16 and VC-E1/12 modules operating in the internal ports (US_NS) mode (see *VC-E1/16 Operating Modes* section on page *1-16*) support the N+1 protection mode, one of the Gmux-2000 APS functions. The N+1 protection function is managed by the Gmux-2000 control subsystem, and operates independently of the other Gmux-2000 APS capabilities.

An N+1 protection group is formed by adding a spare (standby, or protection) module to a group of N modules that carry the traffic (the protected modules). All the N+1 modules must be of the same type, that is, either VC-E1/16 or VC-E1/12.

N+1 protection for VC-E1/16 modules operates as follows:

- When all of the N traffic-carrying (protected) VC-E1/16 or VC-E1/12 modules operate normally, the protection VC-E1/16 module is idle.
- When any one of the N traffic carrying modules reports a malfunction, the malfunctioning module is automatically replaced by the protection module, thereby restoring the original traffic capacity.

  The process used to replace a protected module by the protection module is as follows:

  - The Gmux-2000 control subsystem automatically downloads the configuration of the malfunctioning module to the protection module. Therefore, no configuration data should be prepared by the user for the protection module
  - The Gmux-2000 control subsystem updates the internal traffic flow configuration within the chassis:
    - At the telecom buses side: the malfunctioning module is disconnected from the telecom buses, and the protection module is connected in accordance with the downloaded configuration. Therefore, no change in SDH link mapping takes place
    - At the packet buses side: the Fast Ethernet buses of the malfunctioning module are disconnected from the GbE module(s), and those of the protection module are connected in accordance with the downloaded configuration. Therefore, only the internal packet routing is affected: no change is visible from outside the Gmux-2000 chassis.

  The whole process may require up to 30 seconds, after which service is completely restored.
- After the malfunctioning module is replaced, and is again ready for service, the reverse process automatically takes place, with the result that the protected module is returned to service, and the protection module is again idle and ready to protect any other malfunctioning module in the group.

Gmux-2000 supports a single N+1 protection group per chassis:

- For a Gmux-2000 equipped with a single STM1 module, the maximum number of VC-E1/16 or VC-E1/12 modules that can be protected in this way is 5, with the 6th module serving as the protection module
- For a Gmux-2000 equipped with two STM1 modules, the maximum number of VC-E1/16 or VC-E1/12 modules that can be protected in this way is 4, with the 5th module serving as protection module.

To provide differential quality of service, it is possible to define two protection priority levels, low and high: in the event that protection is required for two modules at the same time, only the high priority module will be protected.

The need for protection switching (*flipping*) is evaluated in accordance with the following criteria, arranged below in decreasing order of severity:

1. Module removed from chassis.
2. Hardware failure.
3. User's flip command (a diagnostic function for APS groups).
4. Priority flip.
5. Link failure.

Therefore, when more than one module need protection at the same time, the protection will be activated only for the module with the more severe criteria, while still observing the user-defined protection priorities. Therefore, separate severity evaluations are made for each protection priority, to select the protected module with the highest severity score, yet a low priority module will get protected only when no protection is needed by a high priority module.

## Voice Compression Subsystem

The transmit path of the voice compression subsystem can convert each payload timeslot to a stream of packets for transmission through the network. The relevant timeslots are selected by the user.

The packets are sent to the main processor, which transfers the packets of each bundle of timeslots either to the appropriate internal data port and subchannel, or to GbE module, via the packet bus interface.

The receive path receives from the main processor streams of packets for each compressed payload timeslot, and sends the decompressed voice streams to the corresponding voice port framer.

## Signaling Processing Subsystem

In the transmit path, the signaling interface retrieves the signaling information of each port from the appropriate timeslots, in accordance with the port signaling transfer mode (CAS or CCS), and transfers it to the main processor for processing.

In the receive path, the signaling interface inserts the signaling information of each port restored by the main processor in the appropriate timeslots.

VC-E1/16 modules do not use the signaling information: the signaling information is only transmitted through the compressed voice connections, for use by the user's equipment at the local and remote endpoints.

### Timing and Clock Generation Subsystem

The timing of each E1 port is derived as follows:

- **Transmit path**: the transmit timing of each external port can be either:
  - Derived from the Gmux-2000 nodal timing. In this case, the equipment connected to the corresponding port must use loopback timing.
  - Locked to the clock signal recovered from the receive line signal of the port. This locks the port transmit timing to the transmit timing of the equipment connected to the port receive clock (loopback timing).

  All the E1 ports, whether data or voice, must use the same clock source.
- **Receive path**: the receive path of each E1 port always receives the clock signal from the external E1 port.

***Note***

*The receive clock frequency must always be equal to the transmit clock frequency.*

### Management Subsystem

The module management subsystem controls the operation of the various circuits located on the VC-E1/16 module, in accordance with the configuration parameters sent by the CONTROL module upon power-up, or after resetting.

The VC-E1/16 configuration parameters are derived from the Gmux-2000 configuration stored on the CONTROL module.

The software necessary for the module operation is stored in a flash memory. The software can also be updated by downloading from the CONTROL module.

## VC-E1/16 Operating Modes

***Note***

*The following description uses the card designations as they appear on supervision terminal and Telnet screens.*

VC-E1/16 can be configured to operate in four modes, which differ with respect to signal flow among the TDM ports: **UE_NE**, **US_NE**, **UE_NS**, and **US_NS**.

The various operating modes are described below. For clarity, the voice compression, signaling interface, and main processor subsystems of *Figure 1-9* have been shown as a single ***voice compression*** **subsystem** in the following figures (*Figure 1-10* to *Figure 1-13*).

Note that in operating mode modes, traffic from the active user ports can also be directed to the packet buses. This connection (from the main processor, via the packet bus interface – see *Figure 1-9*), has been omitted in *Figure 1-10* to *Figure 1-13*, because it is always available, irrespective of the selected mode.

### UE_NE Mode

In this mode, VC-E1/16 accepts user traffic from the 16 or 12 external E1 ports (User External – **UE**) and sends the compressed traffic through the external E1 network ports of the module (Network External – **NE**) (see *Figure 1-10*)

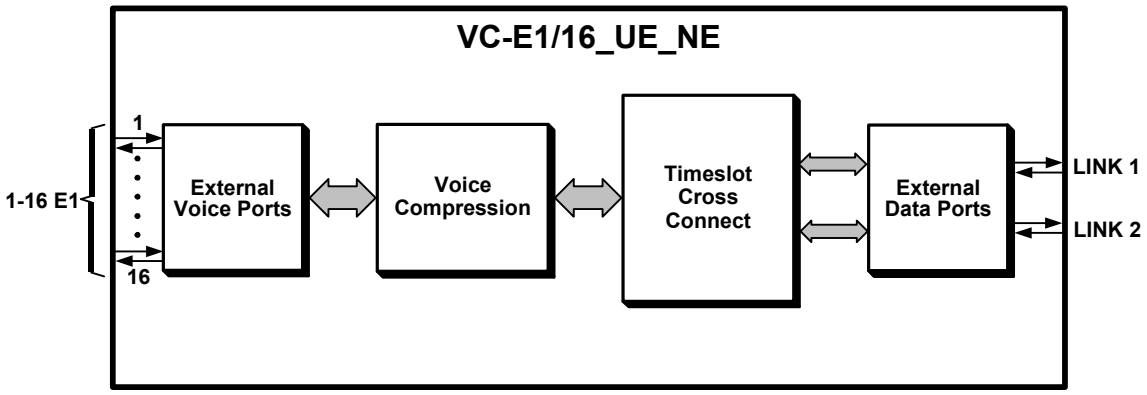



*Figure 1-10. Signal Flow in **UE_NE** Mode*

## US_NE Mode

In this mode, VC-E1/16 accepts user traffic from the SDH ports (User SDH – **US**) located on the STM1 module, and sends the compressed traffic through the external E1 network ports of the module (Network External – **NE**) (see *Figure 1-11*)

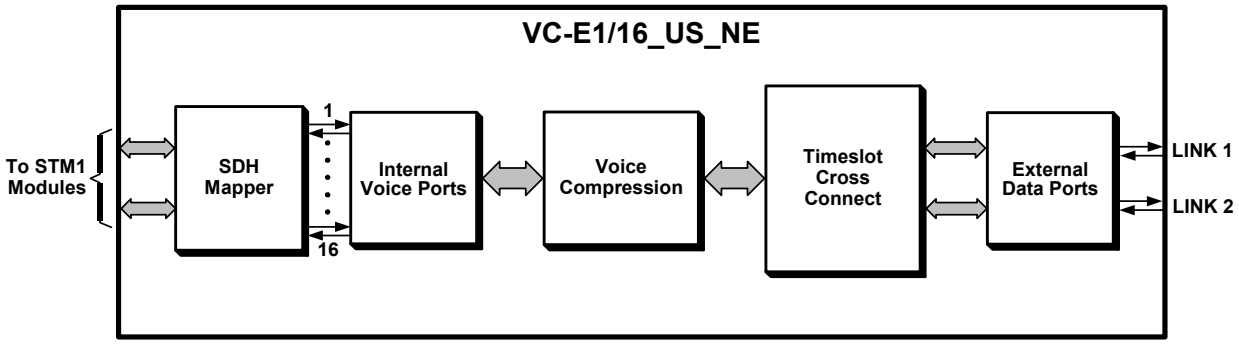



*Figure 1-11. Signal Flow in **US_NE** Mode*

## UE_NS Mode

In this mode, VC-E1/16 accepts user traffic from the 16 or 12 external E1 ports (User External – **UE**), and sends the compressed traffic in VC-12s through SDH ports (Network SDH – **NS**) located on the STM1 module (see *Figure 1-12*)

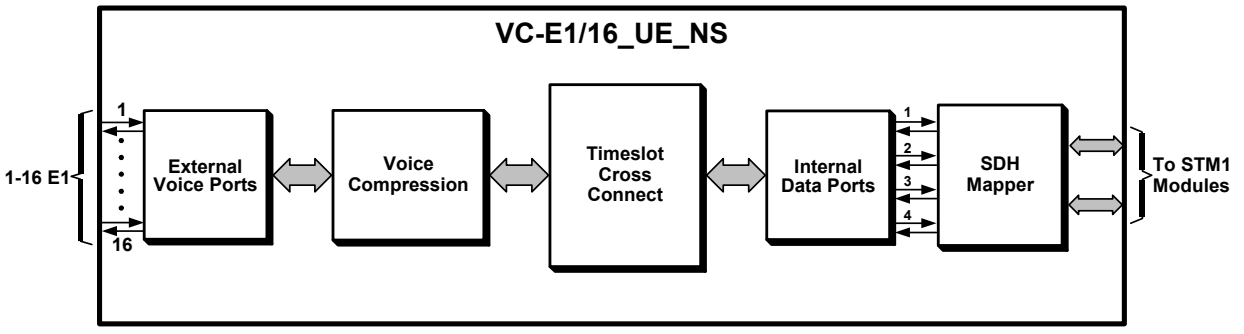



*Figure 1-12. Signal Flow in **UE_NS** Mode*

### US_NS Mode

In this mode, VC-E1/16 accepts user traffic from the SDH ports (User SDH – **US**) located on the STM1 module, and sends the compressed traffic through the same, or other, SDH ports (Network SDH – **NS**) (see *Figure 1-13*).

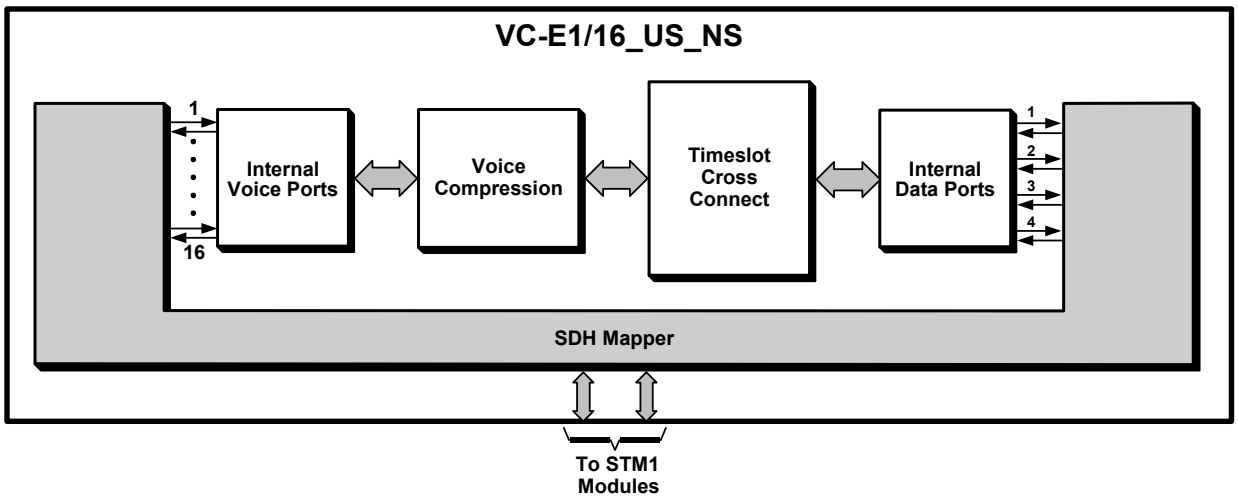



*Figure 1-13. Signal Flow in **US_NS** Mode*

## Processing of Audio Signals

This section explains the processing of audio signals by the VC-E1/16 modules. The processing depends on the signal type. The following functions are available:

- Voice compression
- Processing of inband signaling (DTMF, etc.)
- Processing of fax and voiceband modem signals.

All these functions are implemented by means of advanced digital signal processing (DSP) techniques.

### Handling of Voice Signals

The 64 kbps PCM-encoded voice signals received through the VC-E1/16 user (voice) ports can be compressed using one of the compression algorithms supported by the VC-E1/16 modules. To support voice transmission systems based on both E1 and T1 standards, the user can specify the companding law used by the PCM sections (A-law or μ-law, in accordance with ITU-T Rec. G.711).

The low bit rate voice compression options and data rates supported by VC-E1/16 are as follows:

- Voice compression using multiple-pulse, maximum likelihood code-excited linear prediction (MP-MLQ) per ITU-T Rec. G.723.1, at a channel data rate of 5.3 or 6.4 kbps.
- Voice compression using conjugate structure-algebraic-code-excited linear prediction (CS-ACELP) per Annex A of ITU-T Rec. G.729A, at a channel data rate of 8 kbps.

***Note***

*When you can configure the caller ID signaling protocol (Bellcore Type 1 or V.23), and delay the sending of the caller ID information by a user-selectable interval.*

In addition, the user can also select uncompressed transmission, which requires a channel rate of 64 kbps, by specifying a PCM coder (that is, an ITU-T Rec. G.711 coder with A-law or μ-law companding). This option is not available when using the super-tandem mode.

Although voice is not compressed when using PCM (ITU-T Rec. G.711) coders, all the other signal processing functions and services are active; moreover, you can select a companding law differing from that of the local traffic, to use towards the network.

***Note***

*When using ITU-T Rec. G.711 coders, the total number of active E1 ports per VC-E1/16 is maximum 8.*

With regular voice encoding methods, much bandwidth is wasted during the normal periods of silence in a call (it is often assumed that up to 60% of the call duration consists of quiet intervals). Therefore, to further reduce the actual bandwidth required for voice transmission, the VC-E1/16 modules support voice activity detection (VAD), with silence detection and suppression:

- When a silence interval is detected in a channel (timeslot), an indication is sent to the far end by means of special "silence" packets, that require much less bandwidth than regular voice packets, and the transmitting side releases most of the bandwidth normally occupied by the channel traffic.

  In noisy environments, in particular when other persons carry out conversations near the speaker, the background may sometimes be interpreted as activity on the channel. To improve silence detection under such conditions, two VAD policies, user-selectable for each bundle, are offered:

  - Standard (internal) VAD policy, which complies with the standard silence detection methods
  - RAD proprietary (generic) VAD policy, which allows the user to select the detection threshold: one for high background noise, another for normal background, or to cancel the VAD function altogether. The last selection is not supported for uncompressed (ITU-T Rec. G.711) voice, because in this case the only way to improve bandwidth utilization efficiency is to use the VAD function.
- When decompressing the voice, the far end usually fills the silence interval with noise having characteristics similar to normal background noise (this capability is called comfort noise generation – CNG). This gives the far end subscriber the impression of a live line, and therefore the subjective quality of the call is not noticeably affected. The use of comfort noise generation is also a user-selectable option.

The VAD function enables using less bandwidth to transmit the same amount of voice traffic, without degrading the quality of the call:

- For VC-E1/16, the resulting compression ratio may reach up to 16:1, which means that a single E1 trunk can carry the compressed voice of 16 uncompressed PCM trunks.

- For VC-E1/12, the same compression algorithms are used, but the maximum compression ratio cannot exceed 12:1, because the module has only 12 voice ports.

To improve the perceived communication quality, VC-E1/16 offers additional capabilities:

- Adaptive compensation for delay variations in voice packet transport over the network. The compensation is performed by inserting a jitter buffer; a special mechanism automatically adjusts the jitter buffer size to the actual delay variations, and thus keeps the end-to-end delay in the voice path to the minimum possible.
- Adaptive echo canceling for near-end reflections (echo delay up to 32 msec). The echo canceling performance complies with ITU-T Rec. G.168 requirements, and is user-selectable for each bundle.

  Echoes are generated at the points the transmission path changes from 4-wire to 2-wire. Therefore, echo canceling is needed only when 2-wire equipment (phones, fax, modems) are used at the end points served by this bundle.

  Only one echo canceller should be used at each end point that connects to 2-wire equipment. Therefore, if another echo canceller is already inserted in the signal path of a bundle, for example, at a PBX, the echo canceling function of the VC-E1/16 should be disabled for that bundle.
- Control over the transmit and receive levels of the audio path (the receive path transmits towards the local user's equipment, for example, PBX, and the transmit path receives the signal from the local equipment). This function can be used to compensate for attenuation in the audio paths. For example, when a local subscriber is connected by a long line to the PBX, you can increase the input and output gains to compensate for the expected attenuation, and thus improve the perceived voice quality.

  You can also decrease the receive output level, or increase the nominal transmit input level, as appropriate, to prevent "singing" when echo canceling is not used.

  Being able to select the correct nominal transmit input level (that is, a level that matches the nominal signal level received from the voice equipment) has an additional advantage: it ensures the best voice compression performance, because the compression DSP then operates at its optimum point.
- Support for ***super tandem*** links, that is, links which comprise several segments connected in tandem, where each segment could perform voice decompression and compression. Recompression would significantly degrade voice quality. Therefore, when the ***super tandem*** mode enabled, VC-E1/16 detects whether the arriving voice data has already been compressed by another VC-E1/16 or Vmux unit, and transfers such data unmodified. This is always required on the intermediate segments of tandem links.

  When the ***super tandem*** mode is enabled, the uncompressed voice option (that is, the use of ITU-T Rec. G.711 coders) is not available.

## Processing of Inband Signaling

Inband signaling by means of tones, for example, DTMF, MFR2 and MFC, is widely used in modern telephony networks. Pure tones are also used for various functions, for example, 2000 Hz tones are sometimes used for continuity checks (COT).

The waveform of such signals is very different from speech waveforms, therefore most compression algorithms distort them; for example, distortion of DTMF signals transmitted as analog signals through a compressed voice channel may cause errors to occur in the detection of the dialed digits by PBXs and other equipment.

The method used to overcome this problem is referred to as ***relaying***: when relaying is enabled, the transmit path demodulates the signal to retrieve the encoded data, and transmits the data through the link; the receiving end then synthesizes a clean signal carrying the data, for transmission to the user's equipment. For example, to enable reliable transmission of DTMF signals, the DSP detects incoming DTMF signals, independently for each timeslot, and identifies the dialed digits. The detected digits are digitally transmitted through the link to the far end, where clean digital representations of DTMF signals are synthesized and inserted into the decompressed data stream sent in the corresponding timeslot.

While inband signaling is received, the voice path is disconnected, to prevent interference by signals transmitted through the regular processing path.

The method used for DTMF relaying is also used to transfer transparently call progress tones.

VC-E1/16 modules always relay DTMF signals, using preconfigured parameters that ensure optimal operation.

With respect to 1780 and 2000 Hz tones, and for MF-based signaling (for example, MFR2/R2 and MFC), relaying can be enabled/disabled by the user.

For MF signaling, the user can specify detection parameters, for example, then minimum level and duration a signal must exceed in order to be detected as a signaling code, and can also enable MFC spoofing, which shortens the call setup delay when using R2-MFC signaling by forcing an acknowledge after a user-selectable interval.

***Note***

*Uncompressed voice channels, using PCM (that is, ITU-T Rec. G.711 coders), can transmit inband signaling without significant distortion, yet DTMF and tone relaying are always preferable and should be enabled.*

*MF relaying is always disabled when using an ITU-T Rec. G.711 coder.*

## Automatic Fax Processing

The processing of audio signals by low bit rate voice compression methods does not enable analog transmission of fax signals. Therefore, when it is expected that a compressed voice bundle will carry signals generated by fax machines, it is necessary to enable the automatic fax relaying function.

When automatic fax relaying is enabled, VC-E1/16 will automatically recognize and transmit Group III fax messages at the standard rates in the range of 4.8 to

14.4 kbps. The maximum fax rate can be selected by the user. VC-E1/16 supports automatic fallback capability, that is, it will automatically switch to the next lower data rate supported by both communicating faxes.

The whole fax transmission process is therefore handled as a data transmission, with the VC-E1/16 DSP providing the fax signal modulation/demodulation functions and the detection and generation of the fax connection set up tones (in digital format), to enable the handshaking necessary to implement the standard fax communication protocol. Fax relaying works as follows:

- To set up a fax connection, the DSP processing the local timeslot (channel) emulates the remote fax machine toward the local machine, and the remote DSP emulates the local fax toward the remote machine.
- After the fax connection is established, the fax data stream is transmitted as a packetized data stream through the link. This means that the link must have enough free bandwidth to enable sustained transmission of a data stream at the fax data rate (and the additional connection supervision signals).

This process enables any standard Group III facsimile machine to transmit over the link. The only limitation is that the round-trip transmission delay through the link cannot exceed the time-out intervals specified by the fax communication protocol (about 700 msec); otherwise, the handshaking needed to establish a fax connection will fail.

***Note*** *When necessary, the minimum time a 1100 Hz or 2100 Hz must be present for positive detection of a modem or fax signal can be fine-tuned.*

For bundles expected to carry non-standard fax protocols, which are not recognized by the VC-E1/16 DSPs, the user can configure voiceband fax transmission. In this case, a timeslot carrying a fax transmission is handled as an analog voiceband signal with unknown characteristics, and it is therefore transferred using one of the following types of coders, which are both capable of transferring complex waveforms with little distortion:

- ITU-T Rec. G.711 A-law or μ-law PCM coder (the same coders available for uncompressed voice), which require a channel rate of 64 kbps. PCM requires a bandwidth on the order of one timeslot on the network side
- ITU-T Rec. G.726 ADPCM coders, which compress the signal to a channel rate of 32 kbps (commonly used in Europe, and other countries using the E1-based hierarchy), or 24 kbps (commonly used in North America, and in countries using the T1-based hierarchy). ADPCM provides performance very close to PCM, but requires significantly less bandwidth (40% to 50% of that required by PCM) on the network side.

The selected voiceband coder is used for both fax and modem voiceband transmissions.

## Handling of Voiceband Modem Signals

VC-E1/16 handle voiceband modem signals using either relaying or voiceband signal transmission, as selected by the user:

- Modem relaying is similar to fax relaying, except that the DSP emulates a voiceband modem instead of a fax modem. The recognized modem protocols include V.22/V.22bis, V.32/V.32bis, and V.34 up to 21.6 kbps
- Voiceband transmission, using the selected voiceband coder, can be configured by the user for all the modem calls. When modem relaying is enabled, voiceband transmission is automatically used for any unrecognized modem protocol.

## Handling of Voice Trunk Signaling

Two voice trunk signaling methods are in use, CAS and CCS.

### CAS Handling

Channel-associated signaling (CAS) can be used when an E1 voice port uses G.704 framing with timeslot 16 multiframe. Timeslot 16 carries the signaling information: four signaling bits (designated A, B, C, D) are used for each channel, for a total of 30 signaling channels.

Each VC-E1/16 voice port needs to retrieve the CAS information needed by each channel (timeslot), and transfer it within the voice packets to the far end.

To provide the flexibility needed to adapt to variations to the standard signaling codes that are sometimes implemented by PBX equipment manufacturers, VC-E1/16 supports the definition of specific interpretation, or translation, rules called ***signaling profile***. The profile modifies the signaling information in the direction from the network to the PBX (local user) side.

Each VC-E1/16 voice port can use a different signaling profile. The total number of different signaling profiles that can be defined for each VC-E1/16 is up to 5.

In addition to signaling translation, each profile also enables defining the signaling code to be sent to the local user (PBX) to indicate idle timeslots, and the code sent to indicate the out-of-service state (that is, a timeslot that cannot carry traffic, for example, as a result of a test or maintenance activity).

### CCS Handling

Common-channel signaling (CCS) uses dedicated timeslots within the E1 structure to carry serial data channels for exchanging signaling information between the local and far end. CCS can be used when an E1 voice port uses the basic G.704 framing mode.

With CCS, it is usually necessary to transfer the signaling information transparently through the link, in parallel with the voice payload, and let the end users' equipment interpret the signaling. This transfer uses an optimized HDLC protocol, which is transparent to the endpoints.

For one widely used protocol, Signaling Scheme 7 (SS7), it is possible to increase bandwidth utilization efficiency, because SS7 uses two specific types of protocol signal units (messages) that do not carry signaling information:

- Link status signal units (LSSUs): used to exchange information regarding the status of the SS7 signaling link between two endpoints. After a signaling link

is successfully set up between the SS7 link endpoints, the LSSUs carry little new information.

- Fill-in signal units (FISUs): used just to keep the HDLC signaling link alive. These messages do not carry any information payload, and are transmitted only when there no other types of SS7 signal units to transmit. In the absence of other signaling traffic, the signaling links carry a significant number of FISUs.

To reduce the bandwidth wasted to transmit LSSUs and FISUs, the VC-E1/16 can be configured to suppress a certain fraction of these messages (up to 90%). When the suppression ratio is 0%, the channel performs the same as a HDLC channel.

VC-E1/16 modules support one HDLC timeslot and one SS7 timeslot per voice trunk.

## Packet Processing

The assembly and disassembly of voice packets for transport over the network is handled by the main processor. Two sets of parameters for this purpose are needed:

- Parameters used to define compressed voice (CV) bundles. A bundle consists of selected timeslots from a common voice port that have the same destination, and require identical processing (same voice coding method, same voice processing parameters, etc.).
- Parameters used to define the network connection parameters, that is, a physical port and the assigned bandwidth (number of timeslots) on that port, for the compressed voice traffic. The bandwidth is determined by defining subchannel characteristics.

***Note***

*Multiple bundles may be carried over a single subchannel, provided the available bandwidth is sufficient.*

The voice packets transmitted to the network are inserted in a multiplexed frame structure. The frame can include a total of 100 to 1461 bytes, where a smaller number yields lower delays, and a larger size improves bandwidth utilization efficiency.

A time-out interval (10 to 90 msec) is also defined, after which the current frame is sent even if it is not filled up completely.

### Estimating Bandwidth Requirements

Two basic methods of encapsulation and multiplexing of voice packets are supported by VC-E1/16: TDM for use over IP networks (TDMoIP), and AAL2 for use over MPLS networks (AAL2oMPLS). Both methods take the packets containing the compressed voice and add additional data for enabling transport over the corresponding type of network:

- For TDMoIP, the data includes UDP encapsulation followed by encapsulation in Ethernet frames, with or without VLAN information.

- For AAL2oMPLS, the data includes MPLS encapsulation followed by encapsulation in Ethernet frames, with or without VLAN information.

***Note***

*In addition to traffic packets, VC-E1/16 also sends connectivity check packets.*

The size of the voice packets depends on the selected coding method, and on the encapsulation method.

Since a significant fraction of the number of bytes in each packet is used to carry headers data, VC-E1/16 supports two additional RAD-proprietary, bandwidth-efficient encapsulation methods, identified as TDMoIP+ and AAL2oMPLS+.

The actual number of packets generated depends on the voice activity, and may vary widely over short intervals, although over the long term the average number changes much less.

Despite the statistical variations, it is necessary to make preliminary evaluations of the bandwidth needed to carry the expected traffic. RAD offers a dedicated Bandwidth Calculator, which can be used to predict the required bandwidth, taking into consideration the following main parameters:

- Packet size
- Selected coder
- Silence percentage.

With time, the bandwidth utilization statistics can be analyzed, and the assigned bandwidth be modified to better match actual traffic. For this purpose, VC-E1/16 automatically collects performance statistics, and stores the data in a file, STAT.DAT, that can be sent by TFTP to a RADview-SC/Vmux Service Center.

## Preventing Excessive Bandwidth Consumption

The bandwidth needed to transport compressed voice is not constant, but varies in accordance with the actual amount of intelligence transmitted in each timeslot at each instant. For example, for voice calls the required bandwidth decreases during silence intervals and increases while users talk; in modem or fax calls, the bandwidth is relatively constant for long intervals.

To safely use as much bandwidth on the network connection as possible without degrading the quality of service, the following procedures are used:

- The VC-E1/16 module can signal the equipment connected to voice ports, for example, a PBX, that new calls cannot be accepted when a certain bandwidth utilization threshold is exceeded. For this purpose, VC-E1/16 uses the relevant portions of the protocol specified in ITU-T Rec. Q.50. The threshold at which new calls will be rejected is selectable, and the selection applies to all the voice ports.

  ITU-T Rec. Q.50 specifies a protocol that uses 3 bits in timeslot 16 of the multiframes to establish a communication link between the VC-E1/16 and the equipment connected to each voice port.

  Since this protocol is effective only when the connected equipment supports it, the use of the ITU-T Rec. Q.50 is separately configurable for each voice

port. Moreover, for compatibility with different implementations, the user can select for each voice port the particular implementation:

- ITU-T Rec. Q.50 Annex A: in addition to the communication link in timeslot 16, this Annex enables the user to specify the signaling bits to be used for call setup. Only two bits are used for this purpose: either **A, B** or **C, D**; these bits are selectable separately for each port. VC-E1/16 will ignore the other two bits.
- ITU-T Rec. Q.50 Annex B: this Annex specifies that only signaling bits **A, B** be used for call setup.

- For ongoing calls, VC-E1/16 has an additional mechanism that discards a small portion of voice packets on each call when the available bandwidth is not sufficient to transfer the offered traffic. This mechanism does not depend on support by the connected equipment, and is always active.

  The user can configure the threshold at which voice packets start to be discarded, which must always be higher than the threshold specified for call rejection using ITU-T Rec. Q.50.

  The algorithm used by VC-E1/16 for this purpose leads to a temporary degradation of voice quality, that in many cases is quite tolerable, and is always preferable to call disconnection as a result of congestion. During congestion conditions, buffers may overflow and are flushed. Therefore, overflow event are very disruptive, because they effectively disconnect the link, and any means for avoiding congestion has a very positive effect on voice transmission.

- The last bandwidth control mechanism at the VC-E1/16 level is used to block new modem calls when bandwidth utilization reaches a certain threshold. Since modem calls can be transmitted either as analog (voiceband) signals or using relaying, separate thresholds are used for each type of modem call.

## Processing of Transparent Timeslots

Timeslots configured for transparent transfer bypass the voice compression subsystem, and are converted to packets without any compression.

At the receive side, the data contained in packets carrying the transparent timeslots is directly converted to a data stream that is inserted in the corresponding timeslots, without passing through the voice decompression subsystem. This restores the original payload carried in the selected timeslots.

The user can specify the jitter buffer size used for the data carried by transparent timeslots, in the range of 20 to 100 msec. Since usually such timeslots carry data, this permits to select the most appropriate jitter buffer while restricting the round-trip delay to acceptable values.

Transparent timeslots can be included in each compressed voice bundle. Note that the payload data rate for such timeslots is always equal to the timeslot rate, 64 kbps. The required bandwidth on the links is slightly higher than 64 kbps, because of the packet overhead.

## Diagnostics

To reduce downtime to a minimum, the VC-E1/16 module includes automatic self-test upon power-up and circuits that monitor the operation of the various module subsystems. The self-test results can be read by the Gmux-2000 management subsystem.

Each E1 port has alarm indicators that light in case of local or remote loss of frame synchronization or line signal. In addition, VC-E1/16 supports a remote loopback on each E1 data port, user-activated local and remote loopbacks on each voice port, and local tone injection in selectable voice port timeslots.

The VC-E1/16 collects transmission performance and status data for its various ports, and for the connections (bundles) terminated at the module ports. The collected data enables the system administrator to monitor the transmission performance, and thus the quality of service provided to users, for statistical purposes. In addition, when problems are reported by users served by VC-E1/16, the collected data can also be used for diagnostic purposes, because it can help identify the source of the problem.

# 1.5 Technical Specifications

### General

| | |
|---|---|
| *Function* | Voice compression module |
| *Compression Capacity* | • VC-E1/16: 16 E1 trunks<br>• VC-E1/12: 12 E1 trunks |
| *User-Side Voice Ports* | • 16, respectively 12, external E1 ports<br>or<br>• 16, respectively 12, internal E1 ports connected to SDH ports |
| *Network-Side Data Ports* | • 2 external E1 ports<br>or<br>• 2 internal E1 ports and 2 backup ports, all connected to SDH ports |
| *Packet Switched Network Types* | • UDP over IP<br>• MPLS over ETH |

## User-Side Voice Ports

| | | |
|---|---|---|
| **Voice Port Characteristics** | *Port Type* | E1, 2.048 Mbps |
| | *Compliance* | ITU-T Rec. G.703, G.704, G.706, G.732, G.823 |
| | *Framing* | • Basic G.704 framing, with or without CRC-4<br>• G.704 multiframe, with or without CRC-4 |
| | *Signaling* | • CAS for G.704 multiframe<br>• CCS (transparent transport) for basic G.704 framing |
| | *Jitter Performance* | ITU-T Rec. G.823 |
| **External Voice Port Interface Characteristics** | *Line Interface* | • 120 Ω, balanced<br>• 75 Ω, unbalanced |
| | *Line Code* | HDB3 |
| | *Transmit Level* | • Balanced: ±3V ±10%<br>• Unbalanced: ±2.37V ±10% |
| | *Receive Level* | • DSU mode: 0 to -12 dB<br>• LTU mode: 0 to -36 dB |
| | *Connector (per module)* | 64-pin TELCO female connector |
| **Internal Voice Port Interface Characteristics** | *Interfacing* | VC-12 |
| | *Compliance* | ITU-T Rec. G.783 and G.707 |

## Network-Side Data Ports

| | | |
|---|---|---|
| **Data Port Characteristics** | *Port Type* | E1, 2.048 Mbps |
| | *Compliance* | ITU-T Rec. G.703, G.704, G.706, G.732, G.823 |
| | *Framing* | • Basic G.704 framing, with or without CRC-4<br>• Unframed |

| | | |
|---|---|---|
| | *Bandwidth Utilization* | • Basic G.704 framing: channelized mode with up to 10 independently configurable subchannels per port (1 to 31 timeslots per subchannel)<br>• Unframed: one subchannel, with a capacity of 32 timeslots |
| | *Jitter Performance* | ITU-T Rec. G.823 |
| **External Data Port Characteristics** | *Line Interface* | • 120 Ω balanced<br>• 75 Ω unbalanced (requires adapter cable)<br>User-selected for each port |
| | *Line Code* | HDB3 |
| | *Transmit Level* | • Balanced: ±3V ±10%<br>• Unbalanced: ±2.37V ±10% |
| | *Receive Level* | • DSU mode: 0 to -12 dB<br>• LTU mode: 0 to -36 dB |
| | *Connector* | RJ-45 per port |
| **Internal Data Port Interface Characteristics** | *Interfacing* | VC-12 |
| | *Compliance* | ITU-T Rec. G.783 and G.707 |
| **Backup Characteristics** | *Number of Ports* | • External: 1 primary, 1 secondary<br>• Internal: 2 primary, 2 secondary |
| | *Backup Switching Mode* | Revertive |
| **Packet Bus Interfaces** | | Two Fast Ethernet interfaces to Gmux-2000 internal packet buses |

## Voice Transmission Parameters

| | | |
|---|---|---|
| **Internal Ports** | *Maximum Number* | Up to 32 active bundles per module, each independently configurable |
| | *Bundle Size* | 1 timeslot to full port (30 or 31 timeslots), user-configurable |

| | | |
|---|---|---|
| | *Routing Parameters* | • Selectable source and destination CBID for UDP/IP, or MPLS label for MPLS/ETH<br>• Destination IP address<br>• Next hop IP address<br>• Configurable VLAN support<br>• Selectable exit port and subchannel |
| | *Connectivity Check* | • Transmission rate: 1 to 60 seconds between packets<br>• Alarm threshold: 2 to 5 cycles |
| **Voice Compression (per Bundle)** | *Compression Algorithms* | • MP-MLQ per ITU-T Rec. G.723.1, at rates of 5.3 and 6.4 kbps<br>• Conjugate structure-algebraic-code-excited linear prediction (CS-ACELP) per Annex A of ITU-T Rec. G.729A, at a rate of 8 kbps |
| | *Uncompressed Voice Algorithms* | • ITU-T Rec. G.711 with A-law companding<br>• ITU-T Rec. G.711 with μ-law companding |
| | *Silence Suppression* | • Voice activity detection with silence suppression<br>• Configurable comfort noise generation |
| | *Companding Law* | User-selectable, A-law or μ-law |
| | *Echo Canceling* | ITU-T Rec. G.168, maximum delay 32 msec |
| | *Receive Signal Level* | Adjustable relative to nominal output level: -6 to +6 dB, in 1-dB steps |
| | *Transmit Signal Level* | Adjustable relative to nominal input level: -12 dB, -6 to +6 dB in 1-dB steps, +12 dB |
| **Voice Compression (per Bundle) (Cont.)** | *Encapsulation Method* | For UDP/IP networks:<br>• Standard TDMoIP<br>• RAD proprietary TDMoIP (TDMoIP+)<br>For MPLS networks:<br>• Standard AAL2 over MPLS per ITU-T Y.1414 (AAL2oMPLS)<br>• RAD proprietary AAL2 over MPLS (AAL2oMPLS+) |

| | | |
|---|---|---|
| | *Encapsulation Parameters* | • Configurable packet size (100 to 1461 bytes)<br>• Configurable maximum inter-packet interval (10 to 90 msec) |
| | *Trunk Signaling Support* | • CAS (timeslot 16): user-configurable signaling translation (up to 5 signaling profiles), idle code and OOS code<br>• CCS with user-selectable timeslots: supports ISDN, QSIG, SS7 protocols |
| | *CCS Signaling Support* | • Transparent transfer, using HDLC over AAL2 SSTED per ITU-T Rec. I.366.1<br>• Selectable keep-alive message suppression ratio for SS7 |
| | *Inband Signaling* | • Relaying (detection and generation) of DTMF, MFR2, MFC, complies with EIA/TIA-464B<br>• Configurable MFR2/MFC relaying parameters |
| | *Fax Support* | • Group III fax relay at rates of 4.8 to 14.4 kbps<br>or<br>• Transmission of voiceband fax signals |
| | *Modem Support* | • V.22/V.22bis, V.32/V.32bis, V.34 up to 21.6 kbps<br>or<br>• Transmission of voiceband modem signals |
| | *Voiceband Fax/Modem Coders* | • ITU-T Rec. G.711 with A-law companding<br>• ITU-T Rec. G.711 with μ-law companding<br>• ITU-T Rec. G.726 at 32 kbps<br>• ITU-T Rec. G.726 at 24 kbps |
| **Voice Compression (per Bundle) (Cont.)** | *Bandwidth Utilization Controls* | • Support for ITU-T Rec. Q.50 (Annex A and Annex B, user-selectable), with selectable call rejection threshold. Enabled/disabled on a per-port basis<br>• Random discarding of packets when link bandwidth utilization exceeds selectable threshold. Available on all voice ports<br>• Modem call rejection when link bandwidth utilization exceeds selectable threshold. Configurable on a per-port/per-connection basis |

## Timing

| | | |
|---|---|---|
| **Module Timing** | *E1 Port Transmit Path Timing* | • Loopback timing (transmit timing locked to clock recovered from the received E1 signal)<br>• Gmux-2000 nodal timing |
| | *E1 Port Receive Path Timing* | Locked to clock recovered from the received E1 signal (must be equal to transmit clock rate) |

## General

| | | |
|---|---|---|
| **Module Indicators** | *ACT (green) Indicator* | • Flashes during module initialization and during software downloading.<br>• Lights steadily after initialization is successfully completed, if no fault is detected in the module. |
| | *FLT (red) Indicator* | Hardware fault detected in module |
| **Port Indicators** | *LOC (red) Indicator per Port* | Lights for loss of signal, local loss of synchronization, or local loss of synchronization + AIS |
| | *REM (red) Indicator per Port* | Lights for remote loss of synchronization |
| **Diagnostics** | *Loopbacks* | • Local loopback on each E1 voice port<br>• Remote loopback on each E1 voice port<br>• Local tone injection in selectable voice port timeslots<br>• Remote loopback on each E1 data port |
| | *Self-Test* | Automatically performed upon power-up or reset |
| **Physical** | | Occupies a single I/O slot (I/O-1 to I/O-5, I/O-7 and I/O-9) |
| **Configuration** | | • Programmable via Gmux-2000 management<br>• Fully compatible with the RADview-SC/Vmux Service Center for Vmux Applications |

# Chapter 2

# Installation and Operation

This Chapter provides installation, configuration and operation instructions for the VC-E1/16 module.

The information presented in this Chapter supplements the Gmux-2000 installation, configuration and operation instructions contained in the Gmux-2000 Installation and Operation Manual.

## 2.1 Safety

***Warning***

**Before performing any internal settings, adjustment, maintenance, or repairs, first disconnect all the cables from the module, and then remove the module from the Gmux-2000 enclosure.**
**No internal settings, adjustment, maintenance, and repairs may be performed by either the operator or the user; such activities may be performed only by a skilled technician who is aware of the hazards involved.**
**Always observe standard safety precautions during installation, operation, and maintenance of this product.**

***Caution***

The VC-E1/16 module contains components sensitive to electrostatic discharge (ESD). To prevent ESD damage, always hold the module by its sides, and do not touch the module components or connectors.

***Caution***

To prevent physical damage to the electronic components assembled on the two sides of the module printed circuit boards (PCB) while it is inserted into its chassis slot, support the module while sliding it into position and make sure that its components do not touch the chassis structure, nor other modules.

## 2.2 Installing the Module

VC-E1/16 modules require only one internal setting: the selection of the module operating mode. All the module configuration parameters are determined by software, and therefore there are no additional preparations before installation.

VC-E1/16 modules may be installed in an operating unit (hot insertion).

### Selecting Module Operating Mode

Before installing the module, select its operating mode, that is, the type of voice and network ports that are active. The supported operating modes are described in Chapter 1.

➤ **To select the module operating mode:**

1. Using *Figure 2-1*, identify switch SW2 on the module.
2. Set sections S0 and S1 of SW2 to the positions corresponding to the desired type of ports (use *Table 2-1* as reference). Make sure to leave sections S2 and S3 at OFF.

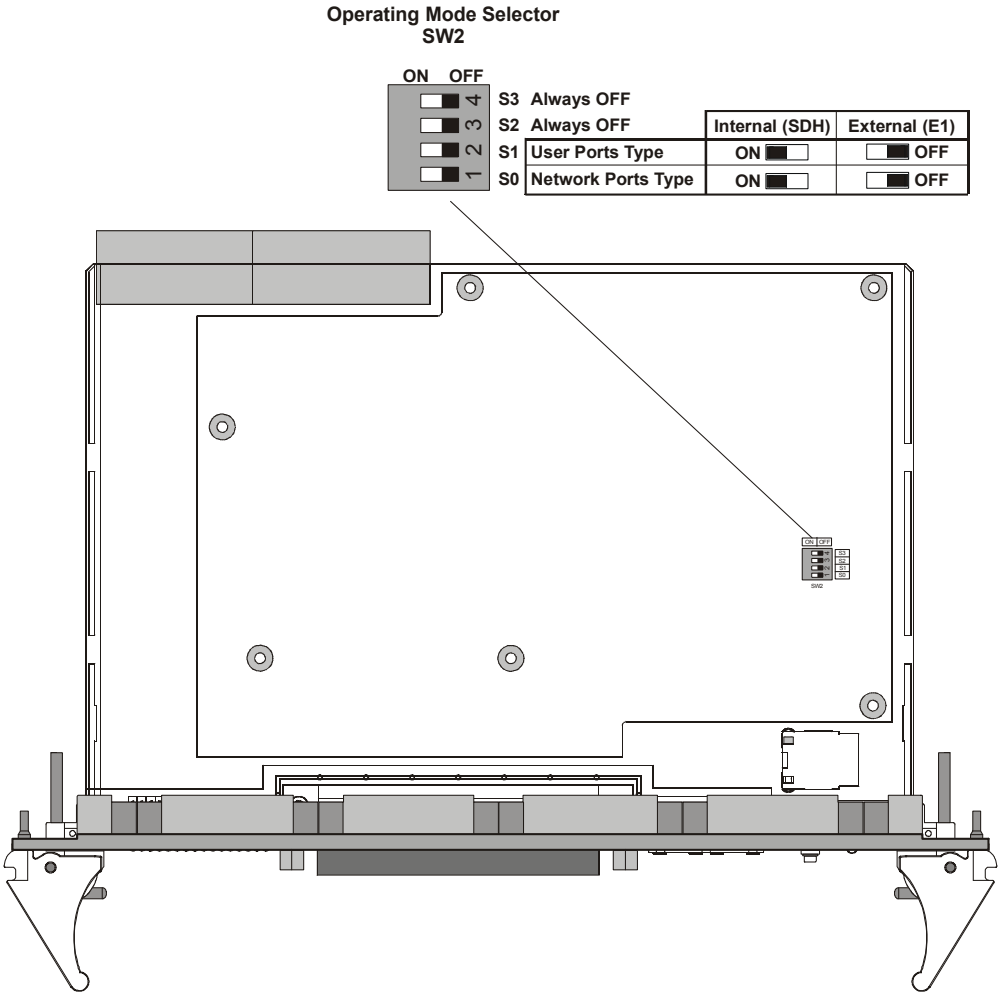
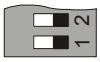
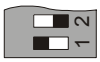
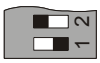
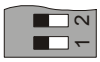



*Figure 2-1. Selecting Module Operating Mode*

*Table 2-1. Selecting VC-E1/16 Operating Mode*

| Card Designation | User-Side (Voice) Ports | Network-Side (Data) Ports | Switch Settings |
|---|---|---|---|
| **VC16E1_UE_NE** | External ports (1-16E1 or 1-12 E1) | External data ports (LINK 1, LINK 2) |  |
| **VC16E1_UE_NS** | External ports (1-16E1 or 1-12 E1) | Internal (SDH) ports |  |
| **VC16E1_US_NE** | Internal (SDH) ports | External data ports (LINK 1, LINK 2) |  |
| **VC16E1_US_NS** | Internal (SDH) ports | Internal (SDH) ports |  |

## Installing a VC-E1/16 Module

➤ **To install a VC-E1/16 module:**

1. Refer to the system installation plan and identify the prescribed module slot: the module can be installed in any one of I/O slots 1 to 5, 7, and 9 of the Gmux-2000 chassis.
2. Check that the fastening screws of the two sides of the module are free to move.
3. Insert the module in its chassis slot, and slide it backward as far as it goes (see *Figure 2-2*).

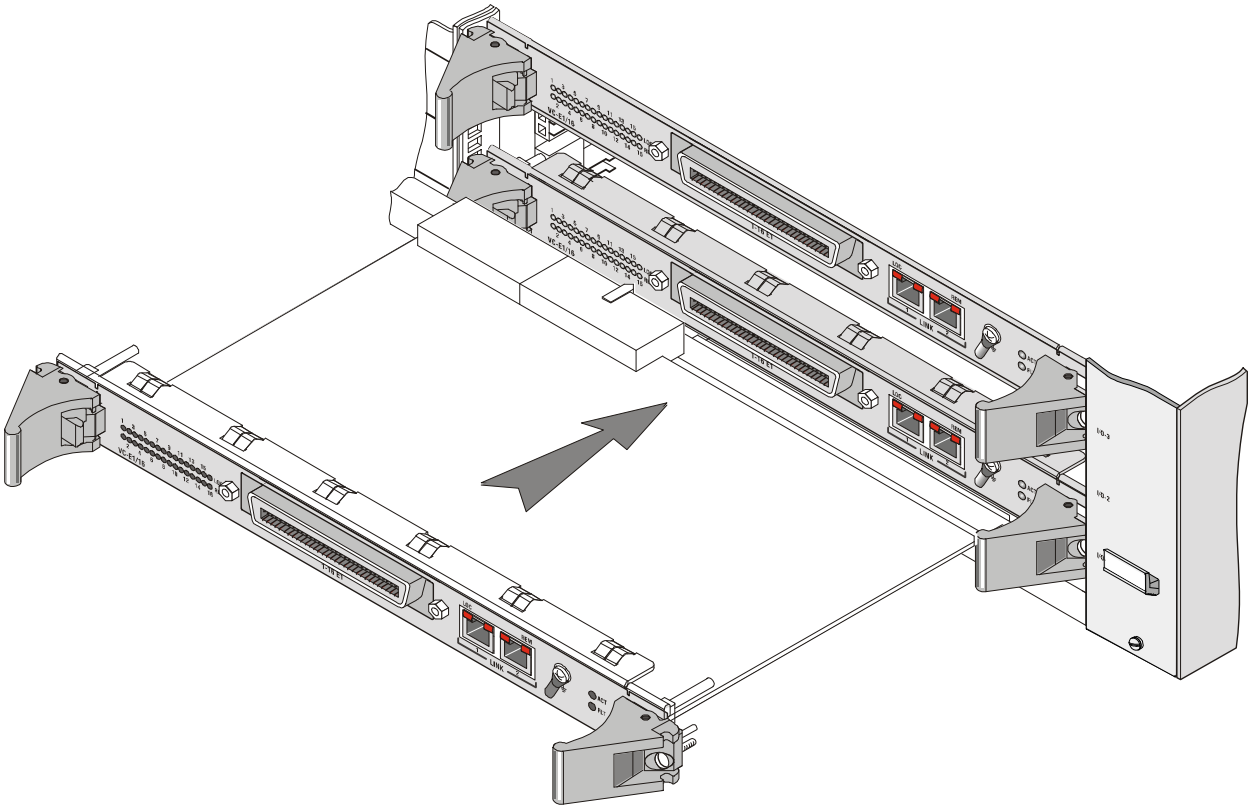


*Figure 2-2. Installing a VC-E1/16 Module*

4. Refer to *Figure 2-3* and press the extractor handles toward the center of the module to fully insert its rear connector into the mating connector on the backplane.
5. Secure the module by tightening its two fastening screws.
6. The module starts operating as soon as it is plugged into an operating enclosure. At this stage, ignore the alarm indications.

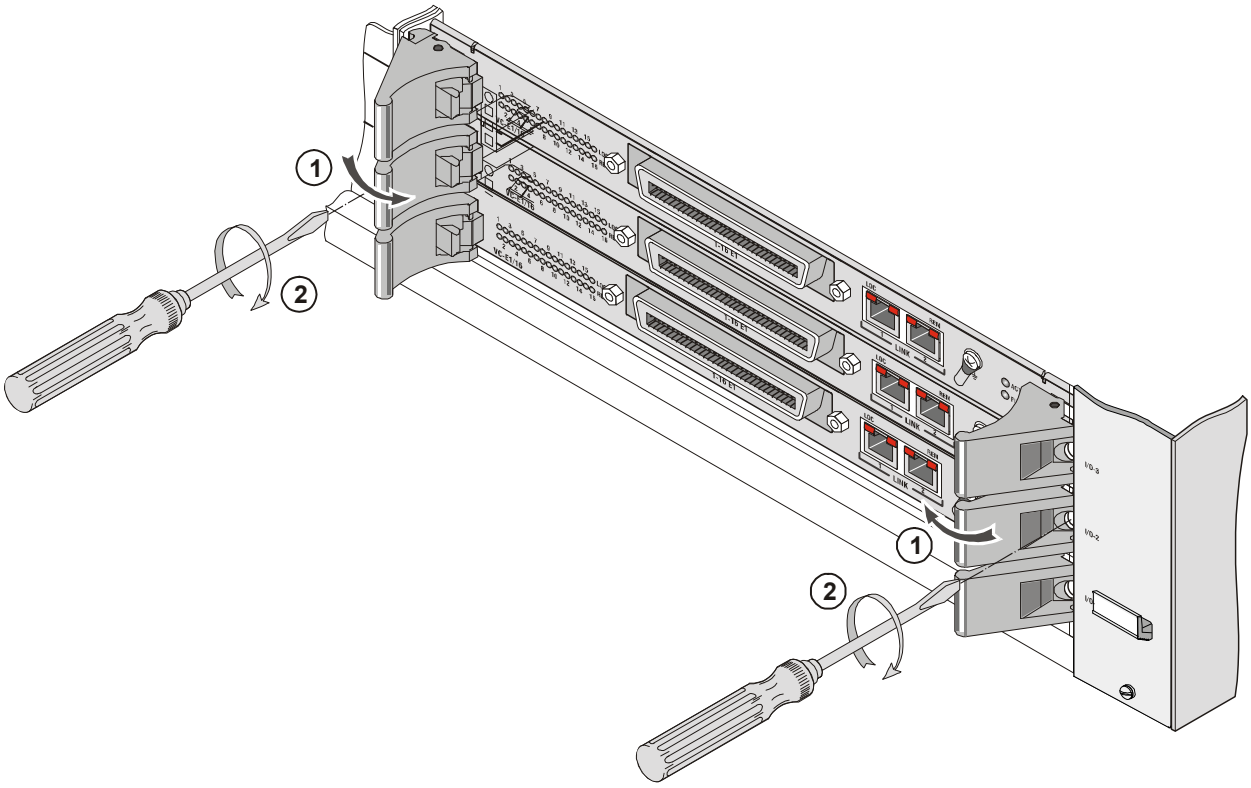



*Figure 2-3. Securing the VC-E1/16 Module to the Chassis*

## Removing a VC-E1/16 Module

➤ **To remove a VC-E1/16 module:**

1. Fully release the two screws fastening the module to the chassis.
2. Press the release buttons of the two extractors (see *Figure 2-4.A*, and ...
3. ... push the extractor handles in the direction shown in *Figure 2-4.B* to disengage the rear connector.
4. Pull the module out.

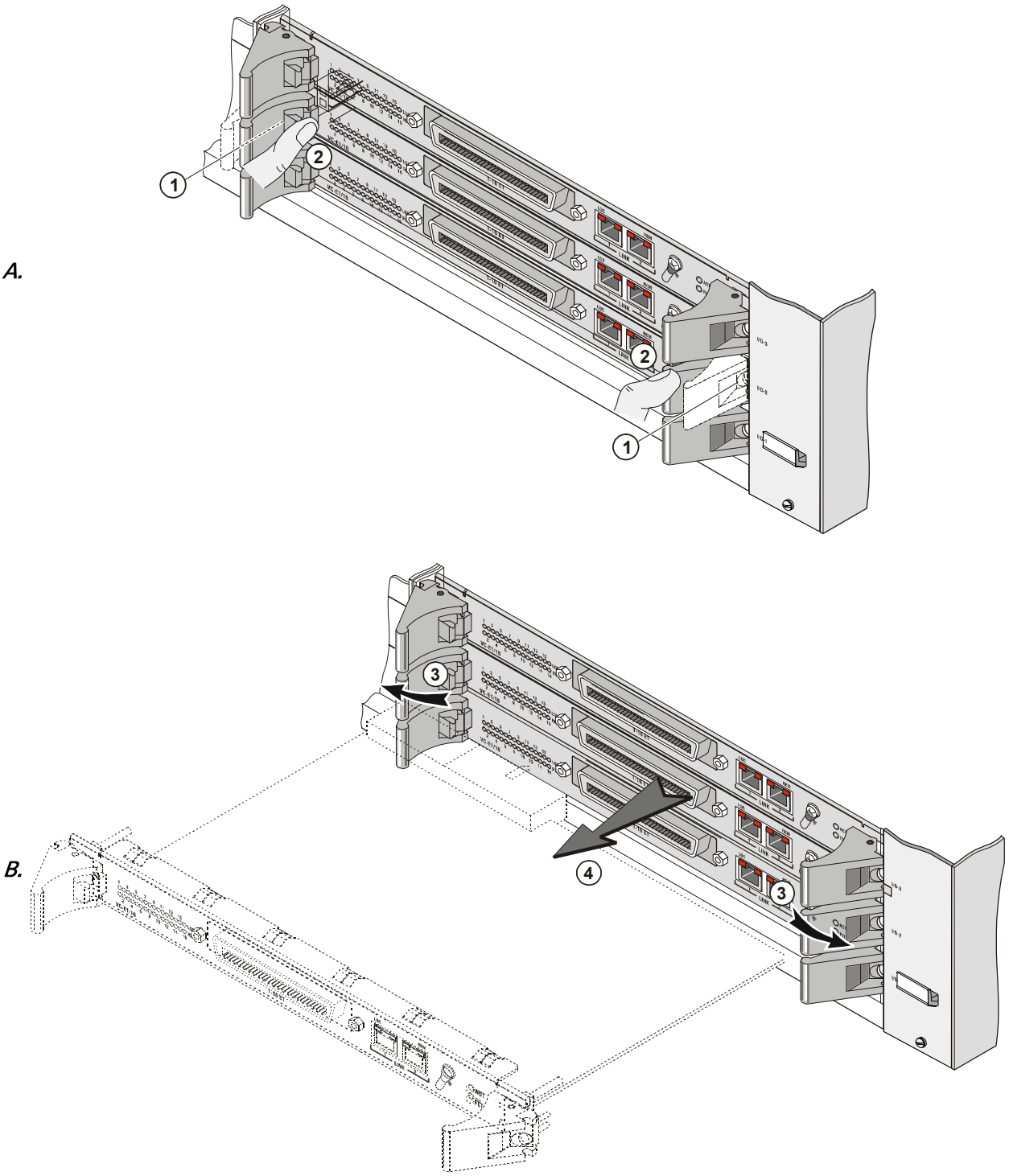



*Figure 2-4. Removing a VC-E1/16 Module*

## 2.3 Connecting Cables

This section provides information regarding connections to the external voice and data ports of the VC-E1/16 module. You may skip this section if the module is connected only to the SDH network.

## Connector Data

### Voice Port Connector

The external E1 voice ports of the VC-E1/16 and VC-E1/12 modules have ITU-T Rec. G.703 interfaces, terminated in one 64-pin D-type female connector, designated 1-16 E1, respectively 1-12 E1. The interface type, balanced or unbalanced, is user-selectable:

- *Table 2-2* lists the pin functions for the 1-16 E1 connector when using the balanced interface.

*Table 2-2. 1-16 E1 Connector, Pin Functions (Balanced Interface)*

| Pin | Function | Pin | Function |
|---|---|---|---|
| 1 | E1 channel 1 transmit output (tip) | 33 | E1 channel 1 transmit output (ring) |
| 2 | E1 channel 1 receive input (tip) | 34 | E1 channel 1 receive input (ring) |
| 3 | E1 channel 2 transmit output (tip) | 35 | E1 channel 2 transmit output (ring) |
| 4 | E1 channel 2 receive input (tip) | 36 | E1 channel 2 receive input (ring) |
| 5 | E1 channel 3 transmit output (tip) | 37 | E1 channel 3 transmit output (ring) |
| 6 | E1 channel 3 receive input (tip) | 38 | E1 channel 3 receive input (ring) |
| 7 | E1 channel 4 transmit output (tip) | 39 | E1 channel 4 transmit output (ring) |
| 8 | E1 channel 4 receive input (tip) | 40 | E1 channel 4 receive input (ring) |
| 9 | E1 channel 5 transmit output (tip) | 41 | E1 channel 5 transmit output (ring) |
| 10 | E1 channel 5 receive input (tip) | 42 | E1 channel 5 receive input (ring) |
| 11 | E1 channel 6 transmit output (tip) | 43 | E1 channel 6 transmit output (ring) |
| 12 | E1 channel 6 receive input (tip) | 44 | E1 channel 6 receive input (ring) |
| 13 | E1 channel 7 transmit output (tip) | 45 | E1 channel 7 transmit output (ring) |
| 14 | E1 channel 7 receive input (tip) | 46 | E1 channel 7 receive input (ring) |
| 15 | E1 channel 8 transmit output (tip) | 47 | E1 channel 8 transmit output (ring) |
| 16 | E1 channel 8 receive input (tip) | 48 | E1 channel 8 receive input (ring) |
| 17 | E1 channel 9 transmit output (tip) | 49 | E1 channel 9 transmit output (ring) |
| 18 | E1 channel 9 receive input (tip) | 50 | E1 channel 9 receive input (ring) |
| 19 | E1 channel 10 transmit output (tip) | 51 | E1 channel 10 transmit output (ring) |
| 20 | E1 channel 10 receive input (tip) | 52 | E1 channel 10 receive input (ring) |
| 21 | E1 channel 11 transmit output (tip) | 53 | E1 channel 11 transmit output (ring) |
| 22 | E1 channel 11 receive input (tip) | 54 | E1 channel 11 receive input (ring) |
| 23 | E1 channel 12 transmit output (tip) | 55 | E1 channel 12 transmit output (ring) |
| 24 | E1 channel 12 receive input (tip) | 56 | E1 channel 12 receive input (ring) |
| 25 | E1 channel 13 transmit output (tip) | 57 | E1 channel 13 transmit output (ring) |
| 26 | E1 channel 13 receive input (tip) | 58 | E1 channel 13 receive input (ring) |
| 27 | E1 channel 14 transmit output (tip) | 59 | E1 channel 14 transmit output (ring) |
| 28 | E1 channel 14 receive input (tip) | 60 | E1 channel 14 receive input (ring) |
| 29 | E1 channel 15 transmit output (tip) | 61 | E1 channel 15 transmit output (ring) |
| 30 | E1 channel 15 receive input (tip) | 62 | E1 channel 15 receive input (ring) |
| 31 | E1 channel 16 transmit output (tip) | 63 | E1 channel 16 transmit output (ring) |
| 32 | E1 channel 16 receive input (tip) | 64 | E1 channel 16 receive input (ring) |

- *Table 2-3* lists the pin functions for the 1-16 E1 connector when using the unbalanced interface.

*Table 2-3. 1-16 E1 Connector, Pin Functions (Unbalanced Interface)*

| Pin | Function | Pin | Function |
|---|---|---|---|
| 1 | E1 channel 1 transmit output (center contact) | 33 | E1 channel 1 transmit output (ground) |
| 2 | E1 channel 1 receive input (center contact) | 34 | E1 channel 1 receive input (ground) |
| 3 | E1 channel 2 transmit output (center contact) | 35 | E1 channel 2 transmit output (ground) |
| 4 | E1 channel 2 receive input (center contact) | 36 | E1 channel 2 receive input (ground) |
| 5 | E1 channel 3 transmit output (center contact) | 37 | E1 channel 3 transmit output (ground) |
| 6 | E1 channel 3 receive input (center contact) | 38 | E1 channel 3 receive input (ground) |
| 7 | E1 channel 4 transmit output (center contact) | 39 | E1 channel 4 transmit output (ground) |
| 8 | E1 channel 4 receive input (center contact) | 40 | E1 channel 4 receive input (ground) |
| 9 | E1 channel 5 transmit output (center contact) | 41 | E1 channel 5 transmit output (ground) |
| 10 | E1 channel 5 receive input (center contact) | 42 | E1 channel 5 receive input (ground) |
| 11 | E1 channel 6 transmit output (center contact) | 43 | E1 channel 6 transmit output (ground) |
| 12 | E1 channel 6 receive input (center contact) | 44 | E1 channel 6 receive input (ground) |
| 13 | E1 channel 7 transmit output (center contact) | 45 | E1 channel 7 transmit output (ground) |
| 14 | E1 channel 7 receive input (center contact) | 46 | E1 channel 7 receive input (ground) |
| 15 | E1 channel 8 transmit output (center contact) | 47 | E1 channel 8 transmit output (ground) |
| 16 | E1 channel 8 receive input (center contact) | 48 | E1 channel 8 receive input (ground) |
| 17 | E1 channel 9 transmit output (center contact) | 49 | E1 channel 9 transmit output (ground) |
| 18 | E1 channel 9 receive input (center contact) | 50 | E1 channel 9 receive input (ground) |
| 19 | E1 channel 10 transmit output (center contact) | 51 | E1 channel 10 transmit output (ground) |
| 20 | E1 channel 10 receive input (center contact) | 52 | E1 channel 10 receive input (ground) |
| 21 | E1 channel 11 transmit output (center contact) | 53 | E1 channel 11 transmit output (ground) |
| 22 | E1 channel 11 receive input (center contact) | 54 | E1 channel 11 receive input (ground) |
| 23 | E1 channel 12 transmit output (center contact) | 55 | E1 channel 12 transmit output (ground) |
| 24 | E1 channel 12 receive input (center contact) | 56 | E1 channel 12 receive input (ground) |
| 25 | E1 channel 13 transmit output (center contact) | 57 | E1 channel 13 transmit output (ground) |
| 26 | E1 channel 13 receive input (center contact) | 58 | E1 channel 13 receive input (ground) |
| 27 | E1 channel 14 transmit output (center contact) | 59 | E1 channel 14 transmit output (ground) |
| 28 | E1 channel 14 receive input (center contact) | 60 | E1 channel 14 receive input (ground) |
| 29 | E1 channel 15 transmit output (center contact) | 61 | E1 channel 15 transmit output (ground) |
| 30 | E1 channel 15 receive input (center contact) | 62 | E1 channel 15 receive input (ground) |
| 31 | E1 channel 16 transmit output (center contact) | 63 | E1 channel 16 transmit output (ground) |
| 32 | E1 channel 16 receive input (center contact) | 64 | E1 channel 16 receive input (ground) |

The 1-12 E1 connector has similar pin functions, except that only the first 12 channels are connected.

RAD offers as optional accessories the following dedicated adapter cables:

- CBL-TELCO-RJ45/16E1, terminated in 16 RJ-45 connectors, is used to connect equipment with standard RJ-45 connectors to the 1-16 E1 and 1-12 E1 connectors. This cable is intended for connection to equipment with balanced interfaces.
- CBL-TELCO-OPEN/2M and CBL-TELCO-OPEN/10M cables, terminated in free leads that can be connected to any terminal block, or to any connector appropriate for your application. These cables can be used for connection to both balanced and unbalanced interfaces.

### LINK E1 Data Port Connectors

The two external E1 data ports of the VC-E1/16 and VC-E1/12 modules have balanced or unbalanced ITU-T Rec. G.703 interfaces. The interface type is software-selectable.

Each LINK port has one RJ-45 eight-pin connector, designated LINK 1 or LINK 2, respectively. The connector is used for both the balanced and unbalanced interface. Connector wiring is listed in *Table 2-4.*

*Table 2-4. LINK Connector, Pin Functions*

| Pin | Function – Balanced Interface | Function – Unbalanced Interface |
|---|---|---|
| 1 | Receive Data input (ring) | Ground |
| 2 | Receive Data input (tip) | Receive Data input |
| 3 | Not connected | Not connected |
| 4 | Transmit Data output (ring) | Ground |
| 5 | Transmit Data output (tip) | Transmit Data output |
| 6 to 8 | Not connected | Not connected |

The connection to equipment with balanced interface is made by any standard RJ-45-to-RJ-45 cable.

In general, when using the unbalanced interface, it is necessary to connect the RJ-45 connector to equipment using BNC connectors. RAD offers as an optional accessory an adapter cable, CBL-RJ45/2BNC/E1/X, with one RJ-45 plug at one end and two BNC female connectors at the other end.

## Adapter Cables

### Adapter Cable CBL-TELCO-RJ45/16E1

Adapter cable CBL-TELCO-RJ45/16E1 is used to connect to the VC-E1/16 voice channels when all the channels use balanced interfaces.

*Figure 2-5* shows the CBL-TELCO-RJ45/16E1 cable structure.

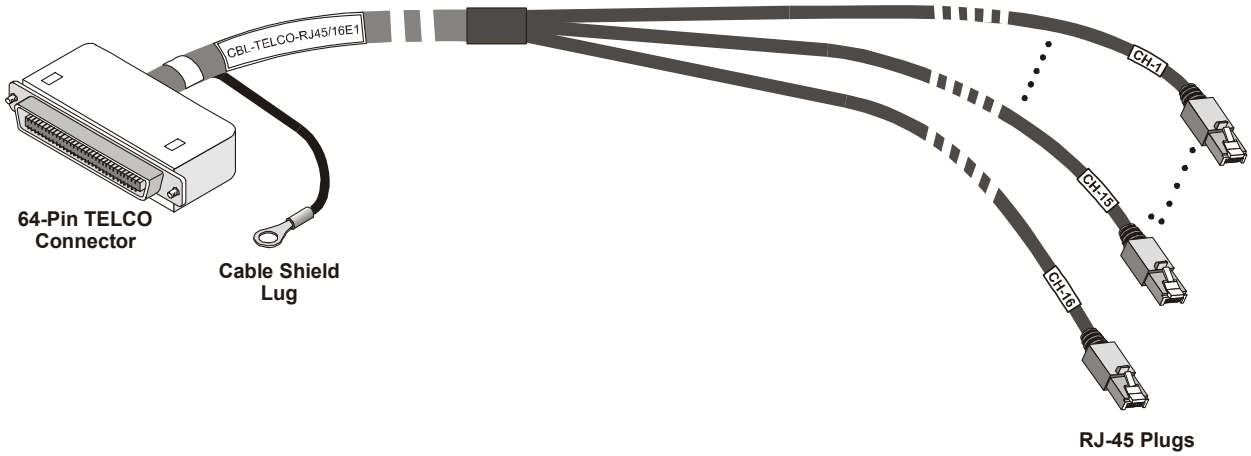



*Figure 2-5. CBL-TELCO-RJ45/16E1 Cable*

The cable, which is 2-meter long, has one 64-pin TELCO male connector that connects to the 1-16 E1 or 1-12 E1 connector of a VC-E1/16, respectively VC-E1/12, module.

The other cable side ends into 16 RJ-45 plugs, designated CH-1 to CH-16. At the 64-pin connector side, the cable shield is terminated in a cable lug, for connection to the dedicated cable shield screw on the module panel.

*Table 2-5* provides the wiring of the CBL-TELCO-RJ45/16E1 cable.

*Table 2-5. CBL-TELCO-RJ45/16E1 Cable Wiring*

| Channel | RJ-45 Connector | 64-Pin Connector | | RJ-45 Connector Pins | Channel | RJ-45 Connector | 64-Pin Connector | | RJ-45 Connector Pins |
|---|---|---|---|---|---|---|---|---|---|
| | | Pin | Function | | | | Pin | Function | |
| 1 | CH-1 | 33 | TX Ring | 1 | 9 | CH-9 | 49 | TX Ring | 1 |
| | | 1 | TX Tip | 2 | | | 17 | TX Tip | 2 |
| | | 34 | RX Ring | 4 | | | 50 | RX Ring | 4 |
| | | 2 | RX Tip | 5 | | | 18 | RX Tip | 5 |
| 2 | CH-2 | 35 | TX Ring | 1 | 10 | CH-10 | 51 | TX Ring | 1 |
| | | 3 | TX Tip | 2 | | | 19 | TX Tip | 2 |
| | | 36 | RX Ring | 4 | | | 52 | RX Ring | 4 |
| | | 4 | RX Tip | 5 | | | 20 | RX Tip | 5 |
| 3 | CH-3 | 37 | TX Ring | 1 | 11 | CH-11 | 53 | TX Ring | 1 |
| | | 5 | TX Tip | 2 | | | 21 | TX Tip | 2 |
| | | 38 | RX Ring | 4 | | | 54 | RX Ring | 4 |
| | | 6 | RX Tip | 5 | | | 22 | RX Tip | 5 |
| 4 | CH-4 | 39 | TX Ring | 1 | 12 | CH-12 | 55 | TX Ring | 1 |
| | | 7 | TX Tip | 2 | | | 23 | TX Tip | 2 |
| | | 40 | RX Ring | 4 | | | 56 | RX Ring | 4 |
| | | 8 | RX Tip | 5 | | | 24 | RX Tip | 5 |
| 5 | CH-5 | 41 | TX Ring | 1 | 13 | CH-13 | 57 | TX Ring | 1 |
| | | 9 | TX Tip | 2 | | | 25 | TX Tip | 2 |
| | | 42 | RX Ring | 4 | | | 58 | RX Ring | 4 |
| | | 10 | RX Tip | 5 | | | 26 | RX Tip | 5 |
| 6 | CH-6 | 43 | TX Ring | 1 | 14 | CH-14 | 59 | TX Ring | 1 |
| | | 11 | TX Tip | 2 | | | 27 | TX Tip | 2 |
| | | 44 | RX Ring | 4 | | | 60 | RX Ring | 4 |
| | | 12 | RX Tip | 5 | | | 28 | RX Tip | 5 |
| 7 | CH-7 | 45 | TX Ring | 1 | 15 | CH-15 | 61 | TX Ring | 1 |
| | | 13 | TX Tip | 2 | | | 29 | TX Tip | 2 |
| | | 46 | RX Ring | 4 | | | 62 | RX Ring | 4 |
| | | 14 | RX Tip | 5 | | | 30 | RX Tip | 5 |
| 8 | CH-8 | 47 | TX Ring | 1 | 16 | CH-16 | 63 | TX Ring | 1 |
| | | 15 | TX Tip | 2 | | | 31 | TX Tip | 2 |
| | | 48 | RX Ring | 4 | | | 64 | RX Ring | 4 |
| | | 16 | RX Tip | 5 | | | 32 | RX Tip | 5 |

## Adapter Cables CBL-TELCO-OPEN/2M, CBL-TELCO-OPEN/10M

The CBL-TELCO-OPEN/2M and CBL-TELCO-OPEN/10M cables are two similar cables, differing only in length: 2 meter/6.5 feet, versus 10 meter/32.8 feet, respectively. The cables can be used to connect to the VC-E1/16 voice ports, irrespective of the selected interface type, balanced or unbalanced.

The cable end connecting to the module is terminated in a 64-pin TELCO male connector, and the other end is terminated in free leads that can be connected to any terminal block, or any connector appropriate for your application.

*Figure 2-6* shows the cable construction, and *Table 2-6* presents the cable wiring and identifies the pair functions for both balanced and unbalanced interfaces.

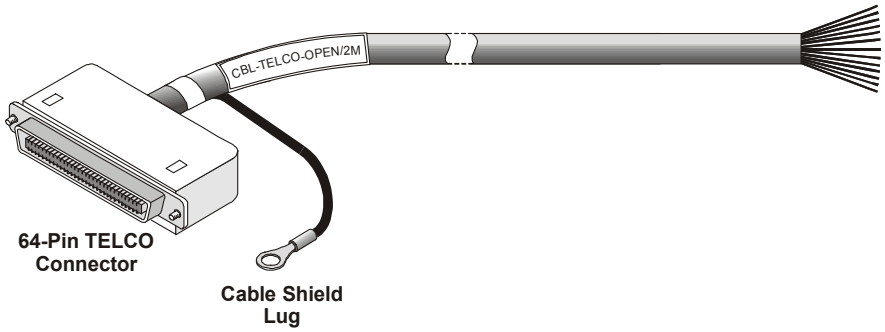



*Figure 2-6. CBL-TELCO-OPEN/2M, CBL-TELCO-OPEN/10M Cables*

*Table 2-6. CBL-TELCO-OPEN/2M Cable Wiring*

| Channel | Function | | Pair Color | Channel | Function | | Pair Color |
|---|---|---|---|---|---|---|---|
| | Balanced | Unbalanced | | | Balanced | Unbalanced | |
| 1 | RX Ring | RX Ground | Orange | 9 | RX Ring | RX Ground | White/Brown |
| | RX Tip | RX Center Contact | White | | RX Tip | RX Center Contact | White |
| | TX Ring | TX Ground | Blue | | TX Ring | TX Ground | Grey/Green |
| | TX Tip | TX Center Contact | White | | TX Tip | TX Center Contact | White |
| 2 | RX Ring | RX Ground | Brown | 10 | RX Ring | RX Ground | White/Grey |
| | RX Tip | RX Center Contact | White | | RX Tip | RX Center Contact | White |
| | TX Ring | TX Ground | Green | | TX Ring | TX Ground | Grey/Brown |
| | TX Tip | TX Center Contact | White | | TX Tip | TX Center Contact | White |
| 3 | RX Ring | RX Ground | White/Blue | 11 | RX Ring | RX Ground | Orange |
| | RX Tip | RX Center Contact | White | | RX Tip | RX Center Contact | Yellow |
| | TX Ring | TX Ground | Grey | | TX Ring | TX Ground | Blue |
| | TX Tip | TX Center Contact | White | | TX Tip | TX Center Contact | Yellow |
| 4 | RX Ring | RX Ground | Green/Blue | 12 | RX Ring | RX Ground | Brown |
| | RX Tip | RX Center Contact | White | | RX Tip | RX Center Contact | Yellow |
| | TX Ring | TX Ground | Orange/Blue | | TX Ring | TX Ground | Green |
| | TX Tip | TX Center Contact | White | | TX Tip | TX Center Contact | Yellow |
| 5 | RX Ring | RX Ground | Grey/Blue | 13 | RX Ring | RX Ground | White/Blue |
| | RX Tip | RX Center Contact | White | | RX Tip | RX Center Contact | Yellow |
| | TX Ring | TX Ground | Brown/Blue | | TX Ring | TX Ground | Grey |
| | TX Tip | TX Center Contact | White | | TX Tip | TX Center Contact | Yellow |
| 6 | RX Ring | RX Ground | Orange/Green | 14 | RX Ring | RX Ground | Green/Blue |
| | RX Tip | RX Center Contact | White | | RX Tip | RX Center Contact | Yellow |
| | TX Ring | TX Ground | White/Orange | | TX Ring | TX Ground | Orange/Blue |
| | TX Tip | TX Center Contact | White | | TX Tip | TX Center Contact | Yellow |
| 7 | RX Ring | RX Ground | Grey/Orange | 15 | RX Ring | RX Ground | Orange/Green |
| | RX Tip | RX Center Contact | White | | RX Tip | RX Center Contact | Yellow |
| | TX Ring | TX Ground | Orange/Brown | | TX Ring | TX Ground | Brown/Blue |
| | TX Tip | TX Center Contact | White | | TX Tip | TX Center Contact | Yellow |
| 8 | RX Ring | RX Ground | Green/Brown | 16 | RX Ring | RX Ground | Grey/Blue |
| | RX Tip | RX Center Contact | White | | RX Tip | RX Center Contact | Yellow |
| | TX Ring | TX Ground | White/Green | | TX Ring | TX Ground | White/Orange |
| | TX Tip | TX Center Contact | White | | TX Tip | TX Center Contact | Yellow |

### Adapter Cable CBL-RJ45/2BNC/E1/X

The 15-cm long adapter cable, CBL-RJ45/2BNC/E1/X, has one RJ-45 plug for connection to the RJ-45 connector, and two BNC female connectors at the other end. Cable wiring is given in *Figure 2-7*.

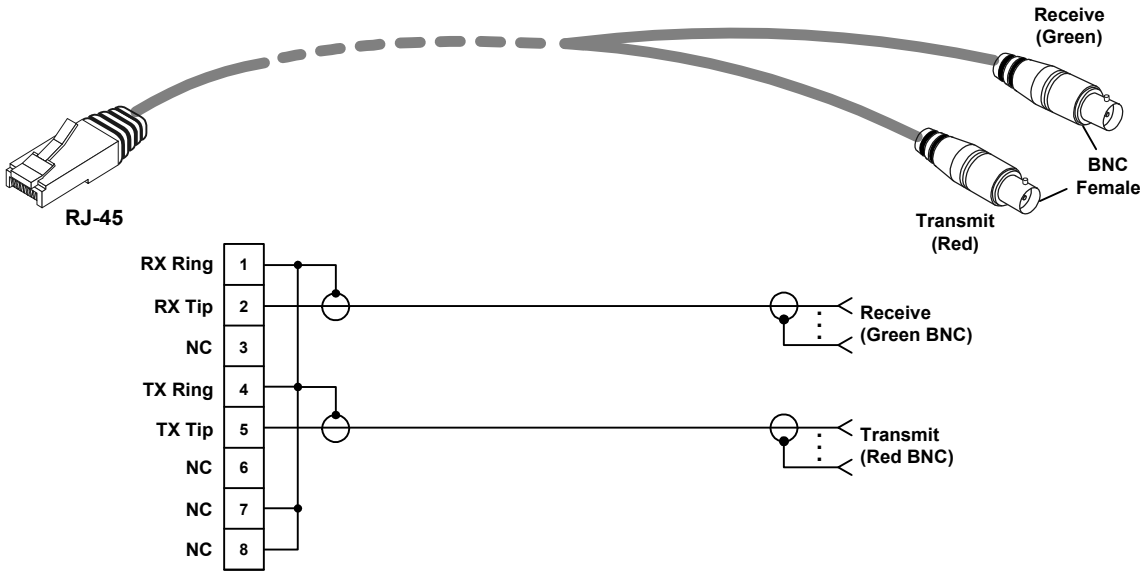



*Figure 2-7. Adapter Cable, CBL-RJ45/2BNC/E1/X, Wiring Diagram*

## Connection Instructions

Before starting, refer to the site installation plan to determine the cables intended for connection to this VC-E1/16 module. Skip the instructions referring to unneeded connections.

### Connecting User's Voice Equipment to the External Voice Ports

***Note***

*Skip this section if the module is configured to use SDH ports for connecting to the user's voice equipment.*

The connections to the E1 user's voice ports of the VC-E1/16 and VC-E1/12 modules are made to the 64-pin connector marked 1-16 E1, respectively 1-12 E1.

➤ **To connect using the CBL-TELCO-RJ45/16E1 adapter cable (balanced interfaces only):**

1. Connect the 64-pin connector of the CBL-TELCO-RJ45/16E1 cable to the module 1-16 E1 or 1-12 E1 connector.
2. Connect the grounding lug of the cable connector to the grounding screw on the VC-E1/16 or VC-E1/12 module panel.
3. Route the cable through the cable guides provided in the rack, to the required location, for example, to an RJ-45 patch panel serving the equipment to be connected to the module.

***Note***

*If the Gmux-2000 is installed in an ETSI rack, it is recommended to route the cable through the Gmux-2000 rack mounting brackets. Make sure to avoid stressing any optical fibers passing through these openings.*

4. Connect each RJ-45 plug (the plugs are marked CH-1 to CH-16) to the prescribed user's equipment or patch panel connector. Insulate unused connectors, to prevent accidental short-circuiting of their exposed contacts to metallic surfaces.

➤ **To connect using the CBL-TELCO-OPEN/2M or CBL-TELCO-OPEN/10M cable (balanced and unbalanced interfaces):**

1. Route the cable through the cable guides provided in the rack to the required location, and then connect its free ends in accordance with the prescribed termination method. See above the *Note* regarding recommended routing.
2. Connect the 64-pin connector of the cable to the module 1-16 E1 or 1-12 E1 connector.
3. Connect the grounding lug of the cable connector to the grounding screw on the VC-E1/16 or VC-E1/12 module panel.

## Connecting to External Data Ports

***Note***

*Skip this section if the VC-E1/16 or VC-E1/12 module is configured to use SDH ports for connecting to the network.*

Each LINK port can use one of the following interfaces (only one can be active at any time):

- 120 Ω balanced interface for operation over twisted pairs
- 75 Ω unbalanced interface for operation over coaxial cable.

Each link port has its own RJ-45 connector:

- To connect to equipment with balanced interfaces, use a standard RJ-45/RJ-45 cable.
- To connect a port to equipment with standard unbalanced interface, use the CBL-RJ45/2BNC/E1/X adapter cable, which has one RJ-45 plug at one end and two BNC female plugs at the other end:
  - Connect the RJ-45 male connector of the adapter cable to the prescribed front panel LINK connector.
  - Connect coaxial cables to the prescribed user's equipment or patch panel connectors. Connect the other end of each cable to one of the adapter cable BNC plugs. Pay attention to correct connection:
    - The red connector is the transmit output of the link port
    - The green connector is the receive input of the link port.

## 2.4 Normal Indications

Upon power up, VC-E1/16 is initialized. During this interval, the ACT indicator flashes at a steady rate, and the FLT indicator must be off.

During normal operation, after initialization is successfully completed:

1. The ACT indicator must light, and the FLT indicator must be off.
2. After the connection to the local and remote equipment is established, all the LOC and REM indicators of the VC-E1/16 module must be off.

# Chapter 3
# Configuration

## 3.1 Introduction

This Chapter provides specific configuration information for VC-E1/16 and VC-E1/12 modules.

The configuration activities are performed by means of the management system used to control the Gmux-2000 unit. The instructions appearing in this Chapter assume that you are familiar with the management system being used:

- Supervision terminal, Telnet or Web browser. Refer to the Gmux-2000 Installation and Operation Manual for instructions.

***Note***

*On the supervision terminal screens, the VC-E1/16 and VC-E1/12 modules are identified as* ***VC16E1****, respectively* ***VC12E1****, or* ***VMX_E1****. See also Table 3-1.*

- Network management system, e.g., the RADview network management system (refer to the RADview User's Manual for instructions).

For general instructions, additional configuration procedures, and background information, refer to the Gmux-2000 Installation and Operation Manual.

## 3.2 Configuration Sequence

***Note***

*If N+1 protection will be used for VC-E1/16 or VC-E1/12 modules, do not prepare configuration data for the protection module in the N+1 APS group.*

To put a VC-E1/16 or VC-E1/12 module into service, you need to perform the following activities:

1. Include a VC-E1/16 or VC-E1/12 module not yet installed in the Gmux-2000 chassis into the database. This allows preconfiguring the module parameters, so that the module will immediately start operating in the desired mode as soon as it is installed in the chassis.

   For the supervision terminal, use **Config>System>Card Type**.

2. Configure the VC-E1/16 global parameters:
   - Signaling profiles for voice ports
   - Bandwidth management parameters.

   For the supervision terminal, use **Config>Physical Layer>I/O**.

3. Configure the E1 voice port parameters. For the supervision terminal, use **Config>Physical Layer>I/O>E1 Port Voice**.
4. Configure the E1 data port parameters, and data port backup configuration. For the supervision terminal, use **Config>Physical Layer>I/O>E1 Port Data**.
5. Configure the parameters related to SDH transport, where applicable (required only when using SDH ports):
   - Configure the VC-12 path parameters for each VC-E1/16 port connected to the SDH network. For the supervision terminal, use **Config>Physical Layer>I/O>VC12 Channel**.
   - Configure the mapping of the VC-12s carrying the VC-E1/16 traffic on the prescribed STM1 module port(s). For the supervision terminal, use **Config>Physical Layer>I/O>Telecom Assignment** for the prescribed STM1 module(s).
6. Configure connection and routing parameters for VC-E1/16 voice ports:
   - Configure connection and routing parameters for each bundle that handles VC-E1/16 voice port traffic. For the supervision terminal, use **Config>Connection**.

***Note***

- *You can configure a bundle using a GbE port as its exit port only after the GbE module has been fully configured, in accordance with the GbE Installation and Operation Manual.*
- *VC-E1/16 uses the host IP address of the CONTROL module as its IP address. Therefore, make sure that the CONTROL module host IP address has already been configured before starting bundle configuration activities (for the supervision terminal, the CONTROL module IP address is configured using **Configuration>System>Host IP**).*

   - Delete all the bundles currently configured for VC-E1/16 voice ports, or disconnect all the bundles from VC-E1/16 voice ports without deleting their configuration data. For the supervision terminal, use **Config>Physical Layer>I/O**.
7. Configure timeslot assignment for the voice and data ports:
   - Configure timeslot assignment for each bundle associated with VC-E1/16 E1 voice ports. For the supervision terminal, use **Config>TS Assignment**.

***Note***

*When a bundle will carry the traffic of a whole E1 voice port, the connection of the port to the bundle can also be made as a part of the E1 voice port configuration.*

   - Configure timeslot assignment for E1 data port subchannels. For the supervision terminal, use **Config>Connection>TS Assignment**.

***Note***

*When the data port operates in the unframed mode, the selection of a subchannel and its connection to the port payload are made as a part of the E1 data port configuration.*

8. When necessary, configure Gmux-2000 for management via VC-E1/16 data ports. Vmux-2100 Voice Trunking Gateways can also be managed in this way.

9. When applicable, configure N+1 protection for VC-E1/16 or VC-E1/12 modules.

   For the supervision terminal, use **Config>System>Redundancy>I/O**.

To display status information and performance monitoring data on the module and its voice and data ports, use the **Monitoring** menu as explained in the Gmux-2000 Installation and Operation Manual.

## 3.3 Including a VC-E1/16 in Gmux-2000 Database

VC-E1/16 has two groups of TDM ports:

- Voice (user-side) ports, identified as ports 1 to 16 (for VC-E1/12, the maximum number is 12)
- Data (network-side) ports, identified as ports 17, 18, or 17 to 20 (the actual number depends on the module operating mode). These ports transfer the processed user's payload to the network.

***Note***

*In addition to the TDM ports, the VC-E1/16 has packet ports that enable using GbE ports as exit ports for the processed user's payload.*

The user can connect each group of TDM ports either to the corresponding external PDH (E1) ports of the module, or to SDH ports located on STM1 modules. When using SDH ports, the connection is made via the Gmux-2000 internal telecom buses; each VC-E1/16 port can be mapped to any desired VC-12.

The TDM port connections depend on the module operating mode, which is selected by means of an internal switch, as explained in *Chapter 2*.

When a new module is programmed in the chassis, it is necessary to manually specify the selected operating mode. For the supervision terminal, the type of ports (external or SDH) each group is connected to is selected by means of the **Config>System>Card Type** screen. This selection must match the configuration selected by means of the internal switch.

***Note***

*Alternatively, the selected module operating mode can be automatically detected by loading the Gmux-2000 hardware configuration, using the **Load HW** command. This option is recommended when starting the configuration of a new VC-E1/16; however, when the database is already prepared, this command may affect the configuration data of other modules, which have been programmed in the database but are not installed in the chassis.*

*Table 3-1* specifies the available operating modes, and identifies the module name to be selected for each mode.

*Table 3-1. VC-E1/16 Operating Modes*

| Card Designation | User-Side (Voice) Ports | Network-Side (Data) Ports |
|---|---|---|
| **VC16E1_UE_NE**, **VC12E1_UE_NE** | External ports (1-16E1 or 1-12E1) | External data ports (LINK 1, LINK 2) |
| **VC16E1_UE_NS**, **VC12E1_UE_NS** | External ports (1-16E1 or 1-12E1) | SDH ports |
| **VC16E1_US_NE**, **VC12E1_US_NE** | SDH ports | External data ports (LINK 1, LINK 2) |
| **VC16E1_US_NS, VC12E1_US_NS** | SDH ports | SDH ports |

➤ **To specify the VC-E1/16 operating mode:**

1. Move the selection block to the prescribed I/O slot. The slots supporting VC-E1/16 modules are I/O-1 to I/O-5, I/O-7 and I/O-9 (a total of seven slots).
2. The modules that can be installed in the selected I/O slot are automatically displayed in a list: select the number corresponding to the desired operating mode, and confirm the selection.

***Note***

*The maximum number of VC-E1/16 modules that can be installed in Gmux-2000 is 7. This maximum number can be installed only when no STM1 module is installed in the chassis (this is always possible when using the **UE_NE** operating mode).*

*However, except for the **UE_NE** operating mode, all the other modes can use ports located on STM1 modules. STM1 modules occupy slots I/O-7 and/or I/O-9. Therefore, the maximum number of VC-E1/16 modules per chassis is 6 when one STM1 module is installed, and 5 when two STM1 modules are installed.*

3. Repeat the procedure for each additional VC-E1/16 module.

## 3.4 Configuring VC-E1/16 Global Parameters

VC-E1/16 global parameters configure processing characteristics that are used in common by all the module voice ports. These characteristics control the interpretation of the signaling information, and the management of the bandwidth required on the link to the network.

***Note***

*If you are not familiar with E1 frame structure, and in particular the utilization of timeslots 0 and 16, which carry framing and signaling information, it is recommended to review Appendix C of the Gmux-2000 Installation and Operation Manual.*

### Configuring Signaling Profiles

Voice calls require the transmission of control (signaling) information, such as call setup and tear-down commands, call status, etc., for each channel.

An often-used method for call control is called channel-associated signaling (CAS). CAS can be used when an E1 port uses G.704 framing with timeslot 16 multiframe (identified as **Framed MF** and **Framed MF – CRC** on supervision terminal screens). In this framing mode, the 32 timeslots of each E1 frame are used as follows:

- Timeslot 0 carries general framing information
- 30 timeslots carry voice channels (the payload itself)
- Timeslot 16 carries the signaling information for all the channels using a special repetitive multiplexed structure spread over 16 frames (this structure is called multiframe). This structure enables using four signaling bits (designated A, B, C, D) for each channel, for a total of 30 signaling channels; the first frame in each 16-frame multiframe carries a multiframe alignment signal that requires 4 bits, an additional bit indicates loss of multiframe alignment, and the remaining 3 bits can be used for other purposes (see the *Bandwidth Management* section below).

To provide the flexibility needed to adapt to variations to the standard signaling codes that are sometimes implemented by PBX equipment manufacturers, VC-E1/16 supports the definition of specific interpretation, or translation, rules called ***signaling profile***. The profile modifies the signaling information in the direction from the network to the PBX (local user) side.

Each VC-E1/16 voice port can use a different signaling profile. The total number of different signaling profiles that can be defined for each VC-E1/16 is up to five.

The definition of signaling profiles is performed by means of the **Signaling Profile Config** item on the **Config>Physical Layer>I/O** screen of the VC-E1/16 module.

The signaling profile itself is specified by means of a table with four columns (one for each signaling bit), identified as **A BIT** to **D BIT**, and three rows:

- **ABCD BITS** – specifies the translation rules for the individual bits
- **IDLE SIGNAL** – specifies the code used for idle timeslots
- **OOS SIGNAL**– specifies the code used to indicate an out-of-service condition.

➤ **To define a signaling profile:**

1. On the **Config>Physical Layer>I/O** screen of the VC-E1/16 module, select **Signaling Profile Config**.
2. On the **Signaling Profile Config** screen, select **Profile Number to Configure**, and then type the desired number, **1** to **5**.

***Note***

*You can also select the number of an existing profile, to modify it.*

3. Now select **Signaling Table Config** to display the profile configuration table for the corresponding signaling profile. The selected **Profile Number** is displayed in the screen header.
4. To specify the translation rules for the signaling bits received from the network, select the bit values in the **ABCD BITS** row. For each bit, you can select one of the following options:
   - **A, B, C, D –** Bit value sent to the local user (PBX) is copied from the corresponding bit received from the network.
   - **NOT A, NOT B, NOT C, NOT D –** Bit value sent to the local user (PBX) is inverted, relative to the value of the corresponding bit received from the network.
   - **0** – Bit value sent to the local user (PBX) is always 0.

- **1** – Bit value sent to the local user (PBX) is always 1.

To select a bit value, move the cursor to the corresponding **BIT** position, and then select the number corresponding to the desired translation.

5. To specify the code sent to the local PBX to indicate idle timeslots, select the bit values in the **IDLE SIGNAL** row.
6. To specify the code sent to the local PBX to indicate the out-of-service state (that is, a timeslot that cannot carry traffic, for example, as a result of a test or maintenance activity), select the bit values in the **OOS SIGNAL** row.

# Bandwidth Management

## Control over Maximum Bandwidth Requirements

The bandwidth needed to transport compressed voice is not constant, but varies in accordance with the actual amount of intelligence transmitted in each timeslot at each instant. For example, for voice calls the required bandwidth decreases during silence intervals and increases while users talk; in modem or fax calls, the bandwidth is relatively constant for long intervals.

To safely use as much bandwidth on the network connection as possible without degrading the quality of service, VC-E1/16 supports three mechanisms:

- The VC-E1/16 module can signal the equipment connected to voice ports, for example, a PBX, that new calls cannot be accepted when network congestion occurs. For this purpose, VC-E1/16 uses the relevant portions of the protocol specified in ITU-T Rec. Q.50. The bandwidth utilization threshold at which new calls will be rejected is selectable, and the selection applies to all the voice ports. The default threshold is 88%, however a higher threshold can be safely selected.

  ITU-T Rec. Q.50 specifies a protocol that uses 3 bits in timeslot 16 of the multiframes to establish a communication link between the VC-E1/16 and the equipment connected to each voice port.

  Since the Q.50 protocol is effective only when the connected equipment supports it, the use of this protocol is separately configurable for each voice port. Moreover, for compatibility with different implementations, the user can select for each voice port the particular implementation:

  - ITU-T Rec. Q.50 Annex A: in addition to the communication link in timeslot 16, this Annex enables the user to specify the signaling bits to be used for call setup. Only two bits are used for this purpose: either **A, B** or **C, D**; these bits are selectable separately for each port. VC-E1/16 will ignore the other two bits.
  - ITU-T Rec. Q.50 Annex B: this Annex specifies that only signaling bits **A, B** be used for call setup.
- For ongoing calls, VC-E1/16 has an additional mechanism that randomly discards a small portion of voice packets on each call when the available bandwidth is not sufficient to transfer the complete traffic. This mechanism does not depend on support by user's equipment, and is always active.

The algorithm used by VC-E1/16 for this purpose leads to a temporary degradation of voice quality that in many cases is quite tolerable, and is always preferable to call disconnection as a result of congestion. The user can configure the threshold at which voice packets start to be discarded. The default threshold is 94%; make sure to select a threshold higher than the threshold specified for call rejection in accordance with ITU-T Rec. Q.50.

- The last bandwidth control mechanism at the VC-E1/16 level is used to block new modem calls when bandwidth utilization reaches a certain threshold. Since modem calls can be transmitted either as analog (voiceband) signals or using relaying, separate thresholds are used for each type of modem call.

  A similar mechanism is supported at the connection (bundle) level.

➤ **To set the bandwidth control thresholds:**

1. On the **Config>Physical Layer>I/O** screen of the VC-E1/16 module, select **Bandwidth Control**.
2. To modify the global bandwidth utilization threshold at which a VC-E1/16 voice port starts rejecting new calls, select **Q.50 Stop New Calls Threshold** and then enter the desired value (range: 80% to 98% of available bandwidth).
3. To modify the global bandwidth utilization threshold at which a VC-E1/16 voice port starts discarding packets on active calls, select **Discard Voice Packets Threshold** and then enter the desired value, in the range of 81% to 99% of the available bandwidth.

***Note*** *The **Discard Voice Packets Threshold** value must always be higher than the **Q.50 Stop New Calls Threshold** value.*

4. To modify the global bandwidth utilization threshold at which a VC-E1/16 voice port starts blocking the set up of new voiceband modem calls, select **Block New Modem VBD Calls Threshold,** and then enter the desired value, in the range of 20 to 98% of the available bandwidth.
5. To modify the global bandwidth utilization threshold at which a VC-E1/16 voice port starts blocking the set up of new modem relay calls, select **Block New Modem Relay Calls Threshold,** and then enter the desired value, in the range of 20 to 98%.

## Reducing Bandwidth Requirements for CCS Signaling

An alternative method for call control is the common channel signaling (CCS). With CCS, one or more timeslots are assigned to transmit the signaling information for all the voice channels carried by the VC-E1/16 port. Timeslot 16 is often used for this purpose, but there are protocols that use other predetermined timeslots, for example, 15, 30, 31.

***Note*** *CCS signaling is relevant only for E1 voice ports using the **Framed G.704** or **Framed G.704 – CRC** framing mode.*

Since there are several CCS protocols in wide use, generally it is necessary to transfer the signaling information transparently through the link, in parallel with

the voice payload. This approach uses the HDLC protocol, and no bandwidth reduction is possible.

However, one widely used protocol, Signaling Scheme 7 (SS7), uses two specific types of protocol signal units (messages) that do not carry signaling information:

- Link status signal units (LSSUs) are used to exchange information regarding the status of the SS7 signaling link between two equipment units. After a signaling link is successfully set up between the SS7 link endpoints, the LSSUs carry little new information.
- Fill-in signal units (FISUs) are used just to keep the HDLC signaling link alive. These messages do not carry any information payload, and are transmitted only when there no other types of SS7 signal units to transmit. In the absence of other signaling traffic, the signaling links carry a significant number of FISUs.

To reduce the bandwidth wasted to transmit LSSUs and FISUs, you can configure the VC-E1/16 to suppress a certain fraction of these messages (up to 90%). When the suppression ratio is 0%, the channel performs the same as a HDLC channel.

When FISU and LSSU suppression is possible, for optimal performance it is recommended to initially select a suppression rate of 30%; latter, the suppression rate can be adjusted as needed.

***Note***

*FISU and LSSU suppression occurs only on timeslots configured for carrying the SS7 protocol (for example, SS7-1). It does not affect timeslots configured as regular HDLC channels for transparent signaling transfer, for example, HDLC-1.*

➤ **To configure the keep-alive suppression rate:**

1. On the **Config>Physical Layer>I/O** screen of the VC-E1/16 module, select **SS7 Keep-Alive Suppression Rate** to display the configuration screen.
2. Select the appropriate fraction of LSSUs and FISUs to be suppressed. The available selections start with 0% (no suppression, also suitable for other protocols), up to 90%, in 10% increments.

## 3.5 Configuring E1 Voice Port Parameters

*Table 3-2* lists all the physical layer configuration parameters for the E1 voice (user side) ports.

➤ **To configure the E1 voice port parameters:**

1. On the **Config>Physical Layer>I/O** screen of the VC-E1/16 module, select **E1 Port Voice**.
2. Select **Channel ID**, and then enter the number of the desired port, 1 to 16 for VC-E1/16, or 1 to 12 for VC-E1/12.

***Note***

*When bundles using ITU-T Rec. G.711 codecs are terminated on the module ports, the maximum total number of active E1 voice ports is 8.*

3. Configure the port parameters in accordance with the prescribed configuration data. Note that some parameters are not applicable when the voice ports are connected to SDH ports.
4. Repeat the procedure for each voice port. You can use the **F** and **B** keys to select the next/previous port.

*Table 3-2. E1 Voice Port Physical Layer Parameters*

| Parameter | Function | Values |
|---|---|---|
| **Channel ID** | Selects the voice port to be configured | **1** to **16** for **VC-E1/16, 1** to **12** for **VC-E1/12.**<br>**Default: 1** |
| **Admin Status** | Used to enable/disable the flow of traffic through the selected port | **NC –** The flow of traffic is disabled. This state should be selected while the port configuration has not yet been completed, or when it is necessary to stop traffic flow through the port.<br>**CONNECTED –** The flow of traffic is enabled.<br>**Default: CONNECTED** |
| **Idle Code** | Selects the code transmitted to fill idle (unused) timeslots in the E1 frames transmitted through this port toward the equipment connected to the port.<br>Make sure to check that a valid idle code is selected. | The available selections are 00 to FF (hexa).<br>**Default: 7F** |
| **Restoration Time** | Used to change the frame synchronization algorithm, to reduce the time required for the port to return to normal operation after local loss of synchronization | **CCITT –** Complies with ITU-T Rec. G.732.<br>**TR-62411 –** Similar to the requirements of AT&T TR-62411 (after 10 seconds).<br>**FAST –** After 1 second.<br>**Default: FAST** |
| **Transmit CLK Source** | Selects the reference source used by the transmit path of this port.<br>Make sure to select the same timing source for all the VC-E1/16 ports | **SYSTEM A –** The port transmit timing is derived from the nodal timing provided by the Gmux-2000 timing subsystem A. In this case, make sure that the equipment connected to the voice port uses loopback timing.<br>**LBT** – The port transmit timing is locked to the clock signal recovered from the receive line signal of the port. This locks the port transmit timing to the transmit timing of the local E1 equipment.<br>**Default: SYSTEM A** |

| Parameter | Function | Values |
|---|---|---|
| **Interface Type** | Selects the characteristics of the line interface, which determine the maximum attenuation of the receive signal that can be compensated for by the port receive path, to obtain the BER performance required by the standards.<br>This parameter is relevant only when using the external port interface | **LTU –** LTU mode, suitable for long haul applications. The maximum supported line attenuation, relative to the nominal transmit level (0 dB), is 36 dB. This selection is not recommended for use with unbalanced interface.<br>**DSU** – DSU mode, suitable for short haul applications. The maximum supported line attenuation, relative to the nominal transmit level (0 dB), is 12 dB. The lower attenuation may actually improve the performance when operating over relatively short line sections, especially when operating over multi-pair cables. In such cables, significant interference is generated by the signals carried by other pairs; when using the **LTU** mode, the interference may be interpreted as a weak signal, and therefore may mask the loss of the desired signal and thus prevent the generation of an alarm.<br>**Default: LTU** |
| **LIU Impedance** | Selects the external port interface.<br>This parameter is not relevant for internal E1 ports (that is, ports connected to SDH ports) | **BALANCE –** Balanced 120 Ω interface, for use over twisted pairs.<br>**UNBALANCE –** unbalanced 75 Ω interface, for use over coaxial cables.<br>**Default: BALANCE** |
| **Profile** | Selects the signaling profile to be used by the corresponding port.<br>The profile specifies the translation of the signaling information received from the network, before being forwarded to the local user's equipment (e.g., a PBX or switch) | The available range is 1 through 5.<br>**Default: 1** |
| **Assign Entire Port to Bundle** | When a single bundle will carry all the timeslots of this port, you can directly select this bundle, instead of using **Config>Connection>TS Assignment.** Make sure to configure the appropriate bundle before trying to select its number | The available selections are 1 to 2000.<br>The selection is made on a separate bundle selection screen, which opens when you try to configure this parameter.<br>**Default: 0** (no bundle selected) |

| Parameter | Function | Values |
|---|---|---|
| **Line Type** | Selects the framing mode of the corresponding port.<br>When all the 31 timeslots of a port are used as voice ports, always select **FRAMED G.704** or **FRAMED G.704 - CRC** | **FRAMED G.704 –** The port handles the traffic as a basic G.704 framed signal, with the CRC-4 option disabled. This mode supports CCS.<br>**FRAMED G.704 - CRC –** The port handles the traffic as a basic G.704 framed signal, with the CRC-4 option enabled. This mode supports CCS.<br>**FRAMED-MF –** The port handles the traffic as a G.704 framed signal with multiframe structure based on timeslot 16, with the CRC-4 option disabled. This mode supports CAS.<br>**FRAMED MF - CRC –** The port handles the traffic as a G.704 framed signal with multiframe structure based on timeslot 16, with the CRC-4 option enabled. This mode supports CAS.<br>**Default: FRAMED-MF** |
| **Q.50 Protocol** | Controls the use of the ITU-T Rec. Q.50 protocol for the selected port.<br>You should enable the protocol only when the equipment connected to the port, for example, PBX, supports this protocol as well.<br>This parameter is relevant only when the port **Line Type** is **FRAMED-MF** or **FRAMED MF - CRC** | **DISABLE** – The use of the ITU-T Rec. Q.50 protocol is disabled.<br>**ANNEX A** – The port uses Annex A of ITU-T Rec. Q.50.<br>**ANNEX B** – The port uses Annex B of ITU-T Rec. Q.50.<br>**Default: DISABLE** |
| **Q.50 Bits Pair** | When the use of ITU-T Rec. Q.50 per Annex A is enabled, selects the pair of bits that carry signaling information.<br>This parameter is displayed only when **Q.50 Protocol** is **ANNEX A** | **A&B** – Signaling carried by bits A and B.<br>**C&D** – Signaling carried by bits C and D.<br>**Default: A&B** |

## 3.6 Configuring E1 Data Port Parameters

The E1 data (network side) ports configuration activities include two tasks:

- Configuration of E1 data port physical parameters
- Configuration of backup parameters for the network-side connection.

### Configuring E1 Data Port Physical Parameters

➤ **To configure the E1 data port parameters:**

1. On the **Config>Physical Layer>I/O** screen of the VC-E1/16 module, select **E1 Port Data**.

2. Select **Channel ID**, and then enter the number of the desired port. The range is 1, 2 for a module using the external data ports (**NE**), and 1 to 4 for a module connecting to SDH ports (**NS**).
3. Configure the port parameters in accordance with the prescribed configuration data. Note that some parameters are not applicable when the data ports are connected to SDH ports.
4. Repeat the procedure for each data port. You can use the **F** and **B** keys to select the next/previous port.

## Configuration Parameters for Framed Modes

*Table 3-3* lists the physical layer configuration parameters for the E1 data ports, when the backup function is disabled, and a framed mode is selected.

*Table 3-3. E1 Data Port Physical Layer Parameters – Backup Disabled*

| Parameter | Function | Values |
|---|---|---|
| **Channel ID** | Selects the data port to be configured | **1** or **2** for a module using the external data ports, and **1** to **4** for a module connecting to SDH ports.<br>**Default: 1** |
| **Admin Status** | See *Table 3-2* | |
| **Backup Status** | Controls the use of backup ports for the connection to the | **ENABLE** – Backup is enabled. See the *Configuring Backup for the Network-Side Connection* section for details.<br>**DISABLE** – Backup is disabled.<br>**Default: DISABLE** |
| **Idle Code** | Selects the code transmitted to fill idle (unused) timeslots in the E1 frames transmitted through this port toward the equipment connected to the port.<br>Make sure to check that a valid idle code is selected.<br>This parameter is not relevant when the **Line Type** is **UNFRAMED** | The available selections are 00 to FF (hexa).<br>**Default: FF** |
| **Restoration Time** | Used to change the frame synchronization algorithm, to reduce the time required for the port to return to normal operation after local loss of synchronization.<br>This parameter is not relevant when the **Line Type** is **UNFRAMED** | **CCITT –** Complies with ITU-T Rec. G.732.<br>**TR-62411 –** Similar to the requirements of AT&T TR-62411 (after 10 seconds).<br>**FAST –** After 1 second.<br>**Default: FAST** |

| Parameter | Function | Values |
|---|---|---|
| **Transmit CLK Source** | Selects the reference source used by the transmit path of this port.<br>Make sure to select the same timing source for all the ports.<br>However, if a VC-E1/16 data port will serve as master clock source for the Gmux-2000, that port must use loopback (**LBT**) timing, and the other ports should use **SYSTEM A** | **SYSTEM A –** The port transmit timing is derived from the nodal timing provided by the Gmux-2000 timing subsystem A. In this case, make sure that the equipment connected to the voice port uses loopback timing.<br>**LBT** – The port transmit timing is locked to the clock signal recovered from the receive line signal of the port. This locks the port transmit timing to the transmit timing of the local E1 equipment.<br>**Default: SYSTEM A** |
| **Interface Type** | See *Table 3-2* | |
| **Sub Channel Configuration** | Displays the current configuration of the subchannels supported by the data port | Displays a table with the following columns:<br>**CHANNEL NUM** – Subchannel number, in the range of 1 to 10.<br>**FUNCTION** – Displays **DATA RELAY**, meaning that the subchannel transfers data transparently.<br>**CONNECT** – Always displays **YES**.<br>The displayed information cannot be changed |
| **Assign Entire Time Slots to Port** | When you want to use the whole bandwidth of the data port for a single subchannel, you can directly select this subchannel and connect to it all the port timeslots, instead of using **Config>Connection>TS Assignment.**<br>This parameter is displayed only when **Line Type** is **FRAMED G.704** or **FRAMED G.704 - CRC** | The selection is made on a separate selection screen, which opens when you try to configure this parameter.<br>See *Assigning all the Port Timeslots to a Subchannel* section on *page 3-36* for details |
| **Line Type** | Selects the framing mode of the port.<br>For point-to-multipoint applications, always select a framed mode: this will enable configuring multiple subchannels on the port for fractional utlization of the port bandwidth | **UNFRAMED –** The port handles the traffic as an unframed 2.048 Mbps data stream. Therefore, the whole bandwidth is available for payload (the equivalent of 32 timeslots, versus 31 timeslots, or 1984 kbps, for the framed modes). See *Specific Configuration Parameters for Unframed Mode* section on *page 3-14* for additional details.<br>**FRAMED G.704 –** The port handles the traffic as a basic G.704 framed signal, with the CRC-4 option disabled. This leaves 31 timeslots for payload.<br>**FRAMED G.704 - CRC –** The port handles the traffic as a basic G.704 framed signal, with the CRC-4 option enabled. This leaves 31 timeslots for payload.<br>**Default: FRAMED G.704** |
| **LIU Impedance** | See *Table 3-2* | |

### Specific Configuration Parameters for Unframed Mode

When **Line Type** is **UNFRAMED**, the port supports a single subchannel, which uses the whole port bandwidth. Therefore, the following changes occur in the list of port parameters presented in *Table 3-3*:

- The **Assign Entire Port to Bundle** parameter is replaced by **Sub Channel Number**. You can then select the number of the subchannel, 1 to 10, assigned to the port (only one subchannel is supported per port). This number is used as a formal reference for displaying information that requires you to enter a subchannel number.
- A new field appears: **TS Type**. You can select between **Connected** (normal operation, traffic can flow), and **NC**, which disables traffic flow through the subchannel, and thus through the whole data port.

Make sure to save your changes before exiting (you will be warned to save by a **Save before Proceeding or Lose your Changes** message). The changes are saved to the edit buffer, and therefore will take effect only when updating the database.

## Configuring Backup for the Network-Side Connection

### Overview of Backup Function

To improve service availability, the data ports can be configured to provide backup for the connection to the network.

In this case, the data ports operate in pairs. In each pair, one port serves as the main (primary) port, and the other as backup (secondary) port. At any time, only one port of each pair carries traffic.

As long as the primary port can carry the traffic, it is automatically selected as the active port. In case of a problem on the link or in the port hardware, the traffic is automatically switched to the backup port. Switching is revertive, that is, when the problem is corrected, the traffic is automatically returned to the primary port.

The port pairs are as follows:

- When using the external data ports, one port can be configured as **Primary** port, and the other as **Backup** (**Secondary**). The total bandwidth on the network connection is thus the bandwidth available on a single port (1984 kbps for framed modes, 2048 kbps for the unframed mode).
- When the network connection is through the SDH ports, ports 1 and 2 can serve as **Primary** ports, and the other two ports (3 and 4) as **Backup** (**Secondary**). The user can select the desired backup port for each primary port and freely select the secondary port for each one (for example, port 3 can be configured as secondary port for either port 1 or port 2, but cannot be configured as secondary to port 4). The total bandwidth on the network connection is thus the bandwidth available on two ports.

The maximum number of active ports is always 2. Therefore, to configure the physical parameters of the ports 3 and 4, first you must enable backup and configure these ports as secondary ports for the main ports, 1 and 2.

Since ports 3 and 4 can serve only as secondary ports, they are enabled only when configured as backup for a primary port. For example:

- If port 1 is configured as regular port (no backup), and port 3 is configured as secondary port for port 2, port 4 is automatically closed.
- If port 2 is configured as regular port (no backup), and port 4 is configured as secondary port for port 1, port 3 is automatically closed.

### Configuring the Backup Function

➤ To configure the backup parameters:

1. Configure the physical port parameters for ports 1 and 2. Make sure to select **Connected** for all the ports.
2. Start backup configuration by selecting the port that will serve as the main (**Primary**) port of the first pair.
3. Select **ENABLE** for **Backup Status**. The currently selected port becomes **Primary** port. The port function is indicated by the **Backup Port Type** field, which appears on the screen.
4. A new parameter, **Backup Port**, appears. Selecting this parameter displays a **Backup Port** selection screen, which presents the available alternatives, in accordance with the VC-E1/16 operating mode.
5. When using SDH ports, if necessary, repeat the procedure for the second pair of ports.
6. At this stage, you can configure the physical port parameters for ports 3 and 4 (which serve as secondary ports).

This completes the configuration. When you will select the backup port for configuration, its function (**Secondary**) is indicated by the **Backup Port Type** field, and the number of the other port of the pair (the **Primary** port) is indicated by a **Backup Port** field.

When configuring the backup port parameters, make sure to select parameters compatible with those of the associated primary port.

## 3.7 Configuring Parameters Related to SDH Transport

As listed in *Table 3-1*, VC-E1/16 ports can be connected to SDH ports. In this case, it is necessary to configure the parameters related to SDH transport. This includes two tasks:

- Configuration of VC-12 path parameters for each VC-E1/16 port connected to the SDH network.
- Configure the mapping of the VC-12s carrying the VC-E1/16 traffic on the prescribed STM1 module port(s).

## Configuring VC-12 Path Parameters

➤ **To configure the VC-12 path parameters:**

1. On the **Config>Physical Layer>I/O** screen of the VC-E1/16 module, select **VC12 Channel**.
2. Select **Channel ID**, and then enter the number of the desired port. The depends on the operating mode, that is, the type of ports connected to the SDH network, as listed in *Table 3-4*.
3. Configure the VC-12 path parameters in accordance with the prescribed configuration data.
4. Repeat the procedure for each port connected to the SDH network. You can use the **F** and **B** keys to select the next/previous port.

*Table 3-4* lists the physical layer configuration parameters for **VC12 Channel**.

*Table 3-4. VC-12 Path Parameters*

| Parameter | Function | Values |
|---|---|---|
| **Channel VC12** | Selects the number of the port for which the path parameters will be configured | The range depends on the operating mode:<br>**VC16E1_US_NE** (only the voice ports are connected to the SDH ports)**: 1** to **16** for VC-E1/16**, 1** to **12** for VC-E1/12.<br>**VC16E1_UE_NS** (only the data ports are connected to the SDH ports)**: 1** to **4**.<br>**VC16E1_US_NS** (both voice and data ports are connected to the SDH ports)**: 1** to **16** for VC-E1/16, **1** to **12** for VC-E1/12 for the voice ports, and **17** to **20** for the data ports (corresponding to data ports 1 to 4).<br>**Default: 1** |
| **Connect State** | Controls the connection of the port to the Gmux-2000 telecom buses, which connect the internal traffic to the STM1 modules | **NC –** The port is disconnected from the telecom buses, and therefore the flow of traffic is disabled. This state should be selected as long as the configuration of the corresponding port has not yet been completed, or when it is necessary to stop traffic flow between the port and the telecom buses.<br>**CONNECTED –** The flow of traffic is enabled.<br>**Default: CONNECTED** |
| **AIS & RDI on Signal Label** | Controls the sending of AIS and RDI indications by the port, in case the received signal label is different from the expected signal label | **ENABLE** – AIS and RDI are sent when a signal label mismatch is detected.<br>**DISABLE** – AIS and RDI are not sent when signal label mismatch is detected. Nevertheless, AIS and RDI are still sent in case of LOP (loss of pointer) or ***unequipped signal label*** condition.<br>**Default: DISABLE** |

| Parameter | Function | Values |
|---|---|---|
| **AIS & RDI on Path Trace** | Controls the sending of AIS and RDI indications by the port, in case the received path trace label (carried in STM-1 overhead byte J2) is different from the expected path trace label | **ENABLE** – AIS and RDI are sent in case a signal label mismatch is detected.<br>**DISABLE** – AIS and RDI are not sent when a signal label mismatch is detected. Nevertheless, AIS and RDI are still sent in case a LOP (loss of pointer) or *unequipped signal label* condition is detected.<br>**Default: DISABLE** |
| **J2 Tx Path Trace** | Controls the sending of the path trace label by the selected port | **ENABLE** – The path trace label is sent.<br>**DISABLE** – No path trace label is sent.<br>**Default: DISABLE** |
| **J2 Rx Path Trace** | Controls the checking of the received path trace label by the selected port | **ENABLE** – Path trace label is checked.<br>**DISABLE** – Path trace label is not checked.<br>**Default: DISABLE** |
| **J2 Path Trace** | Specifies the path trace label.<br>This parameter is relevant only when **J2 Rx Path Trace** and/or **J2 Tx Path Trace** is **ENABLE** | Alphanumeric string of 15 characters. If not all of the 15 characters are needed for the prescribed label, make sure to add spaces up to the required number of characters.<br>**Default: Empty string** |

## Mapping the VC-E1/16 Ports to VC-12s

Each VC-E1/16 port with **Connect State** set to **Connected** must be mapped to the prescribed VC-12 on the prescribed STM1 module port(s). Backup (secondary) ports are mapped independently of the primary ports.

For this purpose, use **Config>Physical Layer>I/O>Telecom Assignment** for the prescribed STM1 module(s). For detailed instructions, refer to the Gmux-2000 Installation and Operation Manual.

# 3.8 Configuring Connections for Voice Ports

The VC-E1/16 voice ports traffic is handled in accordance with the connection parameters defined by the user. For a full description of connection parameters, and detailed configuration instructions, refer to the Gmux-2000 Installation and Operation Manual.

In addition to connection configuration tasks, the main VC-E1/16 physical layer configuration screen provides two related utilities:

- Utility for deleting all the bundles currently configured to serve the voice ports of this VC-E1/16 module. It is recommended to delete the bundles before removing the module, and also before preparing new traffic assignments for this module.

➤ **To delete all the bundles currently configured on this module:**

1. Open **Config>Physical Layer>I/O** for the configured module.
2. Select the **Delete all slot's bundles** option.

- Utility for disconnecting all the bundles currently configured to serve the voice ports of this VC-E1/16 module. This is necessary before modifying the parameters of the bundles served by this module. Disconnecting the bundles also stops the traffic served by the module, without removing the bundle configuration from the database.

➤ **To disconnect all the bundles currently configured on this module:**

1. Open **Config>Physical Layer>I/O** for the configured module.
2. Select the **Disconnect all slot's bundles** option.

## Connection Parameters Configuration Sequence

The tasks needed to configure connections and their parameters are started from the **Config>Connection** menu.

The parameters are specified by configuring a bundle. The total number of bundles that can be defined for each VC-E1/16 is 32, and the total number of bundles that may be defined on a whole Gmux-2000 is 2000.

You can also choose whether to activate the bundle configuration data, or just save it in the database without the bundle being connected to any module.

The configuration of a bundle for VC-E1/16 includes the following main activities:

1. Select the bundle structure: the structure is determined by the connection mode and PSN type.
2. Configure the bundle parameters corresponding to the selected structure. These parameters include routing parameters, and general parameters.
3. Configure the voice processing parameters, and the additional parameters needed to support voiceband modems and fax.

***Note***

*The following sections provide guidelines for performing the required configuration activities, assuming that you are familiar with the Gmux-2000. For full descriptions of connection parameters and detailed configuration instructions, refer to the Gmux-2000 Installation and Operation Manual.*

## Preliminary Configuration Steps

➤ **To select the bundle to be configured for VC-E1/16 voice ports:**

1. Open the **Config>Connection**, and then select **Bundle**.

***Note***

*You can modify an existing bundle only if it is not used by any other module. If a bundle is in use, first disconnect it.*

2. You can either modify an existing bundle by entering its number, or select **Add** to define a new bundle. The new bundle number is automatically assigned the next free number.

   The bundle number appears at the top of each configuration screen.

➤ **To select the bundle structure:**

1. Select **Connection Mode**, and then select **TDMoIP (CV)**. This means that you are configuring a TDMoIP bundle for compressed voice. This is the only type of bundle that can be connected to VC-E1/16 ports.
2. Select **PSN Type**, and then select the type of packet-switched network that will be used to transport the bundle payload:
   - **UDP/IP** – UDP over IP.

***Note***

*If you need to know the number of the source UDP port of a bundle, you can find it as follows (all numbers are in hexadecimal notation):*

***UDP Source Port = 2000 + Destination Bundle Number***

*This means that all the UDP ports numbers are higher than 2000 hexa, that is, 8192 decimal.*

   - **MPLS/ETH** – MPLS over Ethernet.
3. Select **Connection Configuration**, and then select **Function** to select the packet structure version. The available options depend on the selected PSN type.
   - The options available for UDP/IP networks are as follows:
     - **TDMoIP** – TDM over IP with standard packet structure. Always select this type for compatibility with other vendors' UDP/IP equipment.
     - **TDMoIP+** – TDM over IP with RAD proprietary packet structure. This proprietary structure, which increases the connection bandwidth utilization efficiency by reducing the number of overhead bytes carried in each packet, can be used only when the equipment at the other end of the link is another VC-E1/16, or a Vmux Voice Trunking Gateway with compatible software version.

     The default for UDP/IP networks is **TDMoIP+**.
   - The options available for MPLS/ETH networks are as follows:
     - **AAL2oMPLS** – AAL2 over MPLS with standard packet structure. Always select this type for compatibility with other vendors' equipment using AAL2 over MPLS.
     - **AAL2oMPLS+** – AAL2 over MPLS with RAD proprietary packet structure. This proprietary structure, which increases the connection bandwidth utilization efficiency by reducing the number of overhead bytes carried in each packet, can be used only when the equipment at the other end of the link is another VC-E1/16, or a Vmux Voice Trunking Gateway with compatible software version.

     The default for MPLS/ETH networks is **AAL2oMPLS+**.

# Configuring Bundle Parameters

## Configuring Bundle Routing Parameters

➤ To configure the bundle routing parameters:

1. Select **Connection Configuration**, and then select **Routing Parameters**.
2. Obtain the prescribed parameters, and then use *Table 3-5* for parameter descriptions and configuration guidelines.

*Table 3-5. Bundle **Routing Parameters***

| Parameter | Function | Values |
|---|---|---|
| **Connection State** | Selects the state of the connection defined by means of this bundle.<br>The maximum number of bundles that may be simultaneously active for a VC-E1/16 is 32 | **DISABLE** – The bundle is not active. You can still configure and save the desired parameters, to prepare the bundle for activation when needed.<br>**ENABLE** – The bundle is active.<br>**Default: DISABLE** |
| **Destination IP** | Specifies the destination IP address of the bundle.<br>Make sure to select a valid address that can be reached through the port configured in the **Exit Port** field, considering the value selected for the **Next Hop** parameter.<br>The VC-E1/16 source IP address is the CONTROL module host IP address, configured by means of the **Configuration>System>Host IP** screen | Type in the desired IP address, using the dotted-quad format (four groups of digits in the range of 0 through 255, separated by periods).<br>**Default: 0.0.0.0** |
| **Next Hop** | Specifies an IP address to which the bundle packets will be sent, to enable reaching the destination IP address. This is usually the address of an IP router port.<br>You need to specify a next-hop IP address only when the destination IP address is not within the IP subnet that can be reached through the port configured in the **Exit Port** field | Type in the next hop IP address using the dotted-quad format.<br>To use the default gateway, leave this field at the default value, **0.0.0.0**.<br>**Default: 0.0.0.0** |

| Parameter | Function | Values |
|---|---|---|
| **Source CBID** | The function depends on the **PSN Type**: | |
| | **UDP/IP:** Selects the local endpoint source circuit bundle identifier (CBID) at the local endpoint (source) of the connection defined by means of this bundle. | The supported range is **1** to **8063**. Each bundle must have a unique source CBID. **0** means that no source CBID has been defined. |
| | **MPLS/ETH:** Selects the inbound MPLS label accepted for this bundle | The supported range is **1** to **8063**. Each bundle must have a unique inbound MPLS label. **0** means that no label has been defined.<br>**Default: 0** |
| **Dest CBID** | The function depends on the **PSN Type**: | |
| | **UDP/IP:** Selects the circuit bundle identifier at the destination endpoint of the connection defined by means of this bundle. | The supported range is **1** to **8063**. **0** means that no source CBID has been defined. |
| | **MPLS/ETH:** Selects the outbound MPLS label accepted for this bundle.<br>Do not reuse the same **Dest CBID** value on bundles terminating at the same destination IP address, and/or using the same VLAN ID (when VLAN tagging is enabled) | The supported range is **1** to **8063**. **0** means that no label has been defined.<br>**Default: 0** |
| **Exit Port** | Select the port through which the traffic generated by this bundle will be sent.<br>You can select either a VC-E1/16 data port, or one of the GbE module ports | **To specify a VC-E1/16 data port:**<br>Enter the I/O slot of the VC-E1/16 module that will use this bundle, and the data port number, **17** or **18** (when using backup, enter only the number of the primary port). Separate the two digits by a slash **/**.<br>**To specify a GbE module port:**<br>Enter the I/O slot (**6** or **8**) and the physical port number (**1** or **2**) of the desired GbE port. Separate the two digits by a slash **/**.<br>**Default: 0/0** |
| **Exit Channel** | When using as exit port a VC-E1/16 data port, this parameter is used to select the data port subchannel through which the traffic generated by this bundle will be sent.<br>This parameter is not relevant for GbE ports | Enter the number of the prescribed subchannel, **1** to **10**.<br>**Default: 1** |

| Parameter | Function | Values |
|---|---|---|
| **TOS** | Specifies the Layer 3 priority assigned to the traffic generated by this bundle.<br>This priority is indicated by the IP type-of-service (ToS) parameter for this bundle. The specified value is inserted in the IP TOS field of the bundle IP packets.<br>When supported by an IP network, the type-of-service parameter is interpreted in accordance with RFC 791 or RFC 2474, as a set of qualitative parameters for the precedence, delay, throughput and delivery reliability to be provided to the IP traffic generated by this bundle. These qualitative parameters may be used by each network that transfers the bundle IP traffic to select specific values for the actual service parameters of the network, to achieve the desired quality of service<br>You can also specify a Layer 2 priority by means of the **VLAN Priority** field, provided **VLAN Tagging** for this bundle is **Enable** | The ToS parameters are selected on a screen that opens after selecting the **TOS** parameter.<br>The current value of the **TOS** parameter (**00** to **FF** hexa, equivalent to 0 to 255 decimal) is displayed at the top of the screen in hexadecimal notation. The displayed **TOS** value is the sum of the parameter values selected for each ToS component, and it is dynamically updated after selecting a value for each component.<br>The available selections for each ToS component are as follows (each is listed from the lowest to the best value):<br>**Precedence:**<br>• Routine (lowest)<br>• Priority<br>• Immediate<br>• Flash<br>• Flash Override<br>• Critical/ECP<br>• Internetwork Control<br>• Network Control (highest)<br>**Delay**:<br>• Normal<br>• Low<br>**Throughput**:<br>• Normal<br>• High<br>**Reliability**:<br>• Normal Reliability<br>• High Reliability<br>**Default: 00** |
| **VLAN Parameters** | Used to configure the use of VLANs for voice packets. VLANs are relevant only when the traffic traverses Ethernet-based networks.<br>Selecting **VLAN Parameters** displays a submenu that includes the following selections:<br>• **VLAN Tagging**: controls the use of VLAN tagging for the traffic generated by this bundle | **ENABLE** – VLAN tagging is enabled.<br>**DISABLE** – VLAN tagging is disabled.<br>**Default: DISABLE** |

| Parameter | Function | Values |
|---|---|---|
| | • **VLAN ID:** when VLAN tagging is enabled, specifies the VLAN ID number used by the bundle traffic sent through this port. When VLAN tagging is disabled, this parameter is not displayed.<br>When you select the VLAN ID for bundles that exit via a GbE module port, make sure that the same VLAN ID is used for all the bundles having the same destination IP address (the GbE module supports only one VLAN per IP address) | The allowed range, in accordance with IEEE 802.3Q, is **1** to **4094**.<br>**0** means that no VLAN ID has been specified.<br>**Default: 0** |
| | • **VLAN Priority:** when VLAN tagging is enabled, specifies the priority assigned to the bundle traffic using the selected VLAN.<br>When VLAN tagging is disabled, this parameter is not displayed | The allowed range in accordance with IEEE 802.3p is **7** (highest priority) to **0** (lowest priority).<br>**Default: 0** |

## Configuring General Bundle Parameters

➤ **To configure the bundle packetizing parameters:**

1. On the **Connection Configuration** screen, select each of the parameters listed in *Table 3-6* and then select the prescribed values.

   For configuration guidelines and considerations, refer to Chapter 1.

*Table 3-6. General Bundle Parameters – Packetizing Parameters*

| Parameter | Function | Values |
|---|---|---|
| **Max Bytes in Multiplexed Frame** | Specifies the total number of bytes (including all the header information) to be inserted in each packet sent to the network.<br>A larger value increases the bandwidth utilization efficiency, but also increases the connection intrinsic latency, in particular when the bundle is configured, by means of of the **TS Assignment** submenu, to carry a small number of timeslots. | The available selections are **100** to **1461**.<br>**Default: 1400** |
| **Transparent Jitter Size** | Specifies the jitter buffer size used for data carried in transparent timeslots | The jitter buffer size is specified as a multiple of 10 msec.<br>The allowed multiplier range is **2** to **10** (equivalent to 20 to 100 msec, in 10-msec steps).<br>**Default: 2 (20 msec)** |

| Parameter | Function | Values |
|---|---|---|
| **Packetizing Interval** | Specifies the maximum time interval before a TDMoIP packet is transmitted on this bundle.<br>If the packetizing interval expires, the currently processed packet is transmitted even if not yet filled with the number of bytes specified by **Max Bytes in Multiplexed Frame**.<br>Make sure to select an interval compatible with the specified maximum number of bytes. | The allowed range is **10** to **90** msec, in 1-msec steps.<br>**Default: 30 msec** |

➤ **To configure the bundle connectivity check parameters:**

1. On the **Connection Configuration** screen, select **Connectivity Parameters** to display the corresponding submenu.
2. Select each of the parameters listed in *Table 3-7* and enter the prescribed values.

*Table 3-7. General Bundle Parameters – Connectivity Check Parameters*

| Parameter | Function | Values |
|---|---|---|
| **Connectivity Packet Rate (Sec)** | Controls the rate at which keep-alive packets are transmitted for this bundle.<br>The keep-alive packets are used to check connectivity with the destination. When loss of communication is detected, VC-E1/16 generates a BUNDLE OOS alarm, and notifies the equipment connected to the voice port by sending the appropriate OOS indication.<br>Keep-alive packets are relatively small, and therefore require a negligible bandwidth. Therefore, the rate can be selected so as to minimize false alarms while reducing the time needed to reliably detect loss of connectivity:<br>• In a network with low transmission delay (for example, when using TDM links), you can configure a higher rate (shorter interval) to ensure earlier detection of a failure<br>• In packet networks, where delays are more common, it is recommended to set the connectivity parameters to their maximum values | The range is one packet every second (**1**), to one packet every **60** seconds.<br>**Default: 60** |

| Parameter | Function | Values |
|---|---|---|
| **Connectivity Timeout Cycles** | Specifies the number of keep-alive transmission cycles within which VC-E1/16 expects to receive at least one reply from the far end.<br>For example, when the **Connectivity Packet Rate** is 60 seconds and the **Connectivity Timeout Cycles** is **3**, a bundle is declared out-of-service if no reply has been received within 180 seconds of the last reply | The available selections are **2** to **5** cycles.<br>**Default: 3** |

## Configuring Bundle Voice Parameters

➤ **To configure the bundle voice processing parameters:**

1. On the **Connection Configuration** screen, select **Voice Parameters** to display the corresponding submenu.
2. Obtain the prescribed parameters, and then use *Table 3-8* for parameter descriptions and configuration guidelines.

*Table 3-8. Bundle Voice Parameters*

| Parameter | Function | Values |
|---|---|---|
| **Coder/Rate (kbps)** | Selects the coder used for this bundle, and the resulting digitized voice data rate.<br>Note the following limitations related to the use of ITU-T Rec. G.711 coders:<br>• G.711 coders cannot be selected when the **Super Tandem** mode is **ENABLE**.<br>• The maximum number of active E1 voice ports when G.711 coders are used is 8.<br>• Unlike the other options, no real compression occurs with **G.711** coders, and the bandwidth needed to transfer a G.711 coded signal is on the order of one timeslot on the exit port.<br>• The only bandwidth reduction is due to the use of voice activity detection (VAD), and therefore when using a G.711 coder you cannot select **OFF** for the **Noise Level for VAD** parameter.<br>• When using a **G.711** coder, the **MF Parameters** options are not displayed. | **G.723.1/6.4 –** Coder complying with ITU-T Rec. G.723.1, operating at 6.4 kbps.<br>**G.723.1/5.3** – Coder complying with ITU-T Rec. G.723.1, operating at 5.3 kbps.<br>**G.729A/8 –** Coder complying with ITU-T Rec. G.729A, operating at 8 kbps.<br>**G.711 (A-Law) –** PCM coder complying with ITU-T Rec. G.711, operates at a data rate of 64 kbps and uses the A companding law (this companding law is commonly used in Europe, and other countries using the E1 hierarchy).<br>**G.711 (u-Law) –** PCM coder complying with ITU-T Rec. G.711, operates at a data rate of 64 kbps, and uses the μ companding law (this companding law is commonly used in North America, and other countries using the T1 hierarchy.)<br>**Default: G.723.1/6.4** |
| **Caller ID Type** | Selects the caller ID signaling protocol | **Bellcore Type 1** – The caller ID is signalled using the Bellcore Type 1 protocol.<br>**V.23** – The caller ID is signalled using the V.23 protocol.<br>**Default: Bellcore Type 1** |

| Parameter | Function | Values |
|---|---|---|
| **Tx Delay for Caller ID** | Specifies the time to wait from the start of the set up process for a new call, before sending the caller ID information for that call | The available range is **0** to **3000** msec.<br>**Default: 400 msec** |
| **Volume to Line** | Controls the gain of the receive audio path (the path transmitting towards the local user's equipment, for example, a PBX or switch) for this bundle.<br>This function can be used to compensate for attenuation in the local audio path. For example, when a subscriber of the local PBX served by the bundle being configured is connected by a long line, you can increase the gain to compensate for the expected audio line attenuation, and thus improve the perceived voice quality. For example:<br>• If the expected line attenuation is 2 dB, and the PBX nominal input level is 0 dBm, increase the gain by selecting +2dB for **Volume to Line**<br>• When "singing" occurs, decrease the gain.<br>You can also adjust the receive gain to compensate for level differences in the path from the far-end equipment to the local equipment: for example, if the far-end signal is too weak, you can add gain at the output. However, the recommended approach to correct such problems is to adjust the far end **Volume from Line** parameter (see below)s.<br>See *Figure 3-1* below for a description of the way VC-E1/16 interprets the **Volume to Line** and **Volume from Line** parameters. | The available selections are -6 to +6 dB, in 1-dB steps.<br>**Default: 0 dB** |

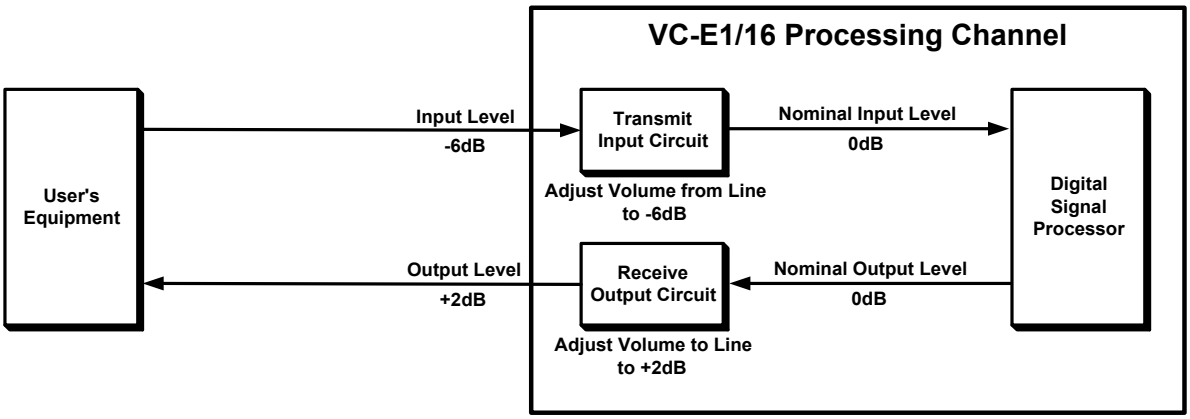



*Figure 3-1. Selection of **Volume to Line** and **Volume from Line** Parameters*

| Parameter | Function | Values |
|---|---|---|
| **Volume from Line** | Controls the nominal input level of the transmit path (the path receiving signals from the local equipment, for example, a PBX) for this bundle.<br>This function is used to compensate for attenuation in the local audio path, or for differences in the nominal levels of the equipment. The purpose is to ensure that the VC-E1/16 DSP processing the signal of this bundle handles a signal at its optimum (nominal) level, and thus can provide the best perceived voice quality. For example:<br>• When the audio signal received from the local PBX in a timeslot served by the bundle being configured is too low, for example, -6B below the nominal level, select -6B for the **Volume from Line** parameter to increase the input gain and thus compensate for the lower input signal.<br>• When the nominal level of the audio signal received from the local PBX is +3 dBm, increase the **Volume from Line** parameter to +3 dB | The available selections are -12 dB, -6 to +6 dB in 1-dB steps, and +12 dB.<br>**Default: 0dB** |
| **Fax/Modem** | Used to control the fax and modem processing parameters for this bundle.<br>These parameters are selected on a separate screen that opens when you select **Fax/Modem**. | |
| | **Modem**<br>Enables/disables modem signal identification for this bundle | **ENABLE** – Modem signal identification is enabled.<br>**DISABLE** – Modem signal identification is disabled.<br>**Default: DISABLE** |

| Parameter | Function | Values |
|---|---|---|
| **Fax/Modem** (cont.) | **Modem Op Mode**<br>Determines the modem signal processing method.<br>This parameter is displayed only when **Modem** is **ENABLE** | **VOICEBAND DATA** – The modem signal is handled as an analog signal. This mode transparently transfers the modem signal waveform, and therefore supports many standards. Note that the bandwidth needed to transfer a voiceband signal requires a relatively large bandwidth on the exit port: the exact bandwidth depends on the selected **VBD Rate** coder, however it is on the order of one timeslot.<br>**RELAY** – The modem signal is demodulated by the receiving side, the resulting data is transmitted through the link as a digital signal, and reconverted to a clean modem signal at the other side. This selection is relevant only for modem signals conforming to one of the standards recognized by the VC-E1/16 (V.22/V.22bis, V.32/V.32bis, and V.34 up to 21.6 kbps): any unrecognized protocol is sent as a voiceband signal. The required bandwidth in this mode is smaller than that required for voiceband signals.<br>**Default: VOICEBAND DATA** |
| | **VBD Rate**<br>Determines the coding method and rate used to transmit voiceband modem signals.<br>Two basic types of coders, which are both capable of transferring complex waveforms with little distortion are used: PCM per ITU-T Rec. G.711, and ADPCM (ITU-T Rec. G.726). ADPCM provides performance very close to PCM, but requires significantly less bandwidth (40 to 50%, depending on version) | **G.711 64k 5 msec –** PCM coder complying with ITU-T Rec. G.711, operating at a data rate of 64 kbps, and sending frames every 5 msec.<br>**G.711 64k 10 msec –** PCM coder complying with ITU-T Rec. G.711, operating at a data rate of 64 kbps, and sending frames every 10 msec.<br>**G.726 32k –** ADPCM coder complying with ITU-T Rec. G.726, operating at a data rate of 32 kbps (this rate is commonly used in Europe, and other countries using the E1-based hierarchy.)<br>**G.726 24k –** ADPCM coder complying with ITU-T Rec. G.726, operating at a data rate of 24 kbps (this rate is commonly used in North America, and other countries using the T1-based hierarchy.)<br>**Default: G.711 64k 5 msec** |

| | | |
|---|---|---|
| **Fax/Modem** (cont.) | **Fax**<br>Enables/disables fax signal relaying for this bundle | **ENABLE** – Fax signal relaying is enabled. The fax signal is identified and demodulated by the receiving side; the resulting data is transmitted through the link as a digital signal, and reconverted to a clean fax signal at the other side. Each end is also capable of interpreting the Group III fax handshaking protocol, to enable setting-up fax calls in accordance with the fax standards.<br>**DISABLE** – Fax signal identification is disabled.<br>**Default: ENABLE** |
| | **Fax Rate (kbps)**<br>Determines the maximum fax transmission rate.<br>When selecting a specific fax rate, you instruct the VC-E1/16 to operate in the fax relay mode: the fax signal is demodulated as a Group III fax signal by the receiving side; the resulting data is transmitted through the link as a digital signal, and reconverted to a clean fax signal at the other side.<br>Each end is also capable of interpreting the Group III fax handshaking protocol, to enable setting-up fax calls in accordance with the fax standards, including support for rate fallback. As a result, all the other standard rates lower than the selected maximum rate are also supported. | **4.8 kbps –** Maximum fax rate of 4.8 kbps.<br>**9.6 kbps –** Maximum fax rate of 9.6 kbps.<br>**14.4 kbps –** Maximum fax rate of 14.4 kbps.<br>**VBD** – The fax signal is handled as an analog signal. This mode transparently transfers the fax modem signal waveform, and therefore supports many standards. Note however that the bandwidth needed to transfer a voiceband signal requires a relatively large bandwidth (on the order of one timeslot) on the exit port.<br>**Default: 14.4 kbps** |
| | **Max. VBD Modem Calls**<br>Determines the bundle bandwidth utilization threshold at which the set up of new voiceband modem calls is blocked.<br>This parameter is displayed only when **Modem** is **ENABLE** | The available range is 1 to 62% of the bundle bandwidth.<br>**Default: 62** |
| | **Max. Relay Modem Calls**<br>Determines the bundle bandwidth utilization threshold at which the set up of new modem calls using the relay mode is blocked.<br>This parameter is displayed only when **Modem** is **ENABLE** | The available range is 1 to 62% of the bundle bandwidth.<br>**Default: 62** |

| | | |
|---|---|---|
| **MF Parameters** | Used to control the MF processing parameters for this bundle.<br>DTMF relaying is always active, irrespective of the selection made for MF signals (relaying means that the DTMF signals are decoded, the corresponding digits and codes are transmitted as data to the other end, where the original DTMF signal is regenerated.<br>The MF signal processing parameters are selected on a separate screen that opens when **MFCR2 RELAY** is **ENABLE.**<br>The **MF Parameters** option is not displayed when the **Coder/Rate** parameter is **G.711 (A-Law)** or **G.711 (u-Law),** because relaying of tones is not necessary for these coders, which transfer complex signals without waveform distortion | **MFCR2 RELAY ENABLE** – MFR2/R2-MFC tone identification is enabled, and thus tone relay is enabled.<br>**MFCR2 RELAY DISABLE** – MF signals identification is disabled.<br>**Default: DISABLE** |
| | **Minimum Pulse Width (msec)**<br>Selects the minimum pulse width of the MF tone that will be accepted for relaying | The supported range is **45** to **300** milliseconds.<br>**Default: 50** |
| | **Minimum Power Level (dBm)**<br>Selects the minimum power of the MF tone that will be accepted for relaying | The supported range is **-1** to **-35** dBm.<br>**Default: -20** |
| | **MFC Spoofing**<br>Controls the use of MFC tone spoofing | **ENABLE** – MFC tone spoofing is enabled. This shortens the call setup delay when using R2-MFC signaling, by forcing an MF Forward Tone acknowledge after the interval selected by means of the **Tone Ack Interval** parameter.<br>**DISABLE** – MFC tone spoofing is disabled.<br>**Default: DISABLE** |
| | **Tone Ack Interval**<br>Determines the tone acknowledge delay.<br>This parameter is displayed only when **MFC Spoofing** is **ENABLE** | The available range is a multiple (**N**) of 20 msec, where **N** is 3 to 20 (equivalent to **60** to **400** msec).<br>**Default: 3 (60 msec)** |
| **Custom Tone Detection** | Enable/disable detection and relaying of tones used by specific types of equipment for this bundle.<br>The selection is made on a separate screen with the following options. | |
| | **Tone Detection**<br>Enable/disable detection and relaying of tones used by specific types of equipment for this bundle | **ENABLE** – Custom tone detection and relaying is enabled.<br>**DISABLE** – Custom tone detection and relaying is disabled.<br>**Default: DISABLE** |

| | | |
|---|---|---|
| **Custom Tone Detection** (cont.) | **Tone Frequency**<br>Selects the detected tones.<br>This parameter is displayed only when **Tone Detection** is **ENABLE** | **2000 Hz** – Detection and relaying of 2000 Hz tones.<br>**1780 Hz** – Detection and relaying of 1780 Hz tones.<br>**2000 Hz + 1780 Hz** – Detection and relaying of both 2000 Hz and 1780 Hz tones.<br>**Default: 2000 Hz** |
| **Super Tandem** | Enable/disable detecting whether voice data arriving on this bundle has already been compressed by another VC-E1/16 or Vmux Voice Trunking Gateway | **ENABLE** – Detection enabled. In this case, the data is not processed (that is, it is not decompressed and recompressed), in order to preserve voice quality.<br>This function is needed at intermediate stations along the bundle transmission path.<br>When **ENABLE** is selected, it is not possible to select **G.711 (A-Law)** or **G.711 (u-Law)** for the **Coder/Rate** parameter.<br>**DISABLE** – Detection disabled. You can safely use this selection when the voice port associated with this bundle cannot receive compressed voice.<br>**Default: DISABLE** |
| **Echo Canceller** | Enable/disable the echo canceller function for this bundle.<br>Echo cancelling is needed only when 2-wire equipment (phones, fax, modems) are used at the end points served by this bundle.<br>Only one echo canceller should be used at each end point that connects to 2-wire equipment. Therefore, if another echo canceller is already inserted in the signal path, for example, at a PBX, disable the echo canceller | **ENABLE** – The echo canceller function is enabled.<br>**DISABLE** – The echo canceller function is disabled.<br>**Default: ENABLE** |
| **Coding** | Selects the voice companding law for this bundle (A-law or μ-law, in accordance with ITU-T Rec. G.711) | **A-Law** – A-law, the standard companding law for E1 trunks.<br>**U-Law** – μ-law, the standard companding law for T1 trunks. You may want to use this setting if the bundle connects to remote equipment, for example, a Vmux Voice Trunking Gateway with T1 interfaces.<br>**Default: A-Law** |

| | | |
|---|---|---|
| **VAD** | Controls the Voice Activity Detection (VAD) mode for this bundle, and selects the noise threshold level.<br>The selection is made on a separate screen with the following options. | |
| | **VAD Method**<br>Controls the use of Voice Activity Detection (VAD) for this bundle | **Generic VAD Policy –** RAD proprietary VAD method, enables selection of noise threshold level.<br>**Internal VAD Policy –** VAD method compatible with the applicable standards.<br>**Default: Generic VAD Policy** |
| | **Noise Level for VAD**<br>Selects the noise threshold for the VAD mechanism.<br>This parameter is displayed only when **VAD Method** is **Generic VAD Policy** | **OFF** – VAD disabled. This option is not available when selecting **G.711 (A-Law)** or **G.711 (u-Law)** for the **Coder/Rate** parameter.<br>**HIGH** – VAD enabled, and the higher noise threshold is selected. This selection is recommended for applications with higher background noise level.<br>**LOW** – VAD enabled, and the lower noise threshold is selected. This is the recommended setting.<br>**Default: LOW** |
| **CNG Mode** | Controls the use of the comfort noise generation (CNG) for for this bundle.<br>The insertion of comfort noise is made at the receiving end, and thus it is effective only when VAD is enabled | **DISABLE** – Comfort noise generation is disabled.<br>**ENABLE** – Comfort noise generation is enabled.<br>**Default: ENABLE** |
| **Advanced Voice Parameters** | Provides access to additional parameters effecting audio signal processing.<br>The selection is made on a separate screen with the following options. | |
| | **Cdis/CONNECTING Detection Time**<br>Select the time needed to detect the presence of 1100 Hz and 2100 Hz tones, before switching to the fax or modem mode | The available selections are multiples of 10 msec, in the range of 10 to 100 msec.<br>**Default: 40** |

# 3.9 Configuring VC-E1/16 Port Timeslot Assignment

With VC-E1/16 modules, timeslot assignment is needed for two purposes:

- To specify which voice port timeslots are served by the bundles configured for VC-E1/16 modules
- When necessary, to select timeslots for transparent transfer and attach them to the desired bundle.
- To specify the timeslots (that is, the bandwidth) assigned to each subchannel (and thus, to the bundles connected to that subchannel) on each VC-E1/16 data port. The number of timeslots determines the transmission bandwidth assigned to each bundle. For bandwidth utilization considerations, refer to Chapter 1.

## Configuring Timeslot Assignment for Voice Ports

The voice port timeslots to be served by a compressed voice bundle can be selected in the following ways:

- Select individual timeslots on the corresponding VC-E1/16 voice port. For the supervision terminal, use **Config>TS Assignment**.
- Select a range of consecutive timeslots on the corresponding VC-E1/16 voice port. For the supervision terminal, use **Config>Connection>Assign TS Range to Bundle**.
- When a bundle will carry the traffic of a whole E1 voice port, the connection of the whole port to the bundle can be initiated either from **Config>Connection>TS Assignment**, or from the E1 voice port physical parameters configuration screen.

### Selecting Individual Timeslots

When you select a specific VC-E1/16 voice port, you are provided with a timeslot map that includes the available timeslots: for example, if the port uses **FRAMED MF** or **FRAMED MF - CRC** framing, you will see only timeslots 1 to 15, and 17 to 31 (timeslot 16 is reserved for CAS). For **FRAMED G.704** or **FRAMED G.704 - CRC** framing, all the 31 timeslots are included in the map.

➤ **To configure individual timeslot assignment:**

1. For each timeslot, you need to select the serving bundle. Move the cursor under the timeslot in the **Bundle** row, and then select **Change Cell** and type the prescribed bundle number, in the range of 1 to 2000. **0** means that the timeslot is not yet connected to a bundle.

***Note***

*The required bundles must be defined before starting timeslot assignment.*

2. For each timeslot, you need to select its type. The type determines the processing of the traffic carried in the timeslot. The selection is made on a list that appears when the cursor is within the **TS Type** row. Move the cursor under the timeslot, and then select **Change Cell** to make a selection:

| | |
|---|---|
| **NC** | Timeslot is not connected to the selected bundle. |
| **VOICE** | Voice timeslot, that is, a timeslot with channel-associated signaling information. |
| **HDLC-1** | Timeslot whose payload is interpreted as HDLC data, for example, a timeslot that carries CCS data. |
| **SS7-1** | Timeslot whose payload is interpreted as being signaling data per Signaling Scheme 7. The **SS7 Keep-Alive Suppression Rate** parameter described in the *Reducing Bandwidth Requirements for CCS Signaling* section applies to this timeslot type. |
| **Trans** | Transparent timeslot, that is, a timeslot whose payload is transferred without any processing to the far end (for example, a timeslot that carries encrypted voice or data). |

***Note*** *As an alternative, you can first use the **Assign Entire Port to Bundle** or **Assign TS Range to Bundle** to preconfigure the timeslots and their types, and then modify only the different timeslots.*

3. When operating in a link with the Vmux Voice Trunking Gateway, you need to specify the destination port and the port timeslot at the Vmux. This is made by moving the cursor to the **Dest Port**, respectively **Dest TS**, rows, and using the procedure described in *Step 1* above to make a selection.

## Assigning a Group of Timeslots to a Bundle

You can select a range of timeslots (i.e., a group of consecutive timeslots) of a VC-E1/16 voice port to one bundle in one step, using **Assign TS Range to Bundle** on the **Config>Connection>TS Assignment** screen.

After this step, you can modify the configuration of any desired timeslots as explained in the *Selecting Individual Timeslots* section above.

➤ **To assign a range of timeslots to a bundle:**

1. Select **Assign TS Range to Bundle** to display the configuration screen.
2. Select **Bundle Number** and enter the prescribed bundle number, in the range of 1 to 2000. **0** means that the timeslot is not yet connected to a bundle.

***Note*** *The required bundles must be defined before starting timeslot assignment.*

3. Select **TS Type** and specify the type assigned to all the timeslots in the specified range, as explained in the *Selecting Individual Timeslots* section above.
4. When operating in a link with the Vmux Voice Trunking Gateway, you need to specify the destination port and the port timeslot at the Vmux. This is made by selecting **Dest Port** row, and entering the prescribed Vmux port number.

***Note***

*Each local timeslot assigned to the selected bundle is automatically connected to the far-end timeslot with the same number.*

5. Select **From Ts** and specify the first timeslot in the group to be assigned. Make sure to select a number that allows to include in the bundle the required number of timeslots, as defined by means of **Num of TS**: for example, if the port uses **FRAMED MF** or **FRAMED MF - CRC** framing and you want to connect to the bundle 20 timeslots, the highest timeslot number that can be selected is 10 (considering that timeslot 16 is assigned to CAS signaling).
6. Select **Num of TS** and enter the number of timeslots to be connected to the selected bundle.
7. Confirm the connection by selecting **Connect the TS Range to Bundle.**

## Assigning a Whole Port to a Bundle

You can select all the timeslots of a VC-E1/16 voice port to one bundle in one step, using **Assign Entire Port to Bundle** on the **Config>Connection>TS Assignment** screen, or on the corresponding **Config>Physical Layer>I/O>E1 Port Voice** screen.

After this step, you can modify the configuration of any desired timeslots as explained in the *Selecting Individual Timeslots* section above.

➤ **To assign all the port timeslots:**

1. Select **Assign Entire Port to Bundle** to display the configuration screen.
2. Select **Bundle Number** and enter the prescribed bundle number.
3. Select **TS Type** and specify the type assigned to all the port timeslots.
4. Select **Dest Port** and enter the prescribed far end port number.

***Note***

*Each local timeslot is automatically connected to the far-end timeslot with the same number.*

5. Confirm the connection by selecting **Connect Entire Port to Bundle.**

## Disconnecting a Bundle from a Port

You can disconnect all the timeslots of a VC-E1/16 voice port that are connected to a specific bundle in one step, using one of the procedures described above but with one difference:

- To disconnect a range of timeslots from the bundle: instead of selecting **Connect the TS Range to Bundle**, select **Disconnect the TS Range from Bundle**.
- To disconnect all the port timeslots from the bundle: instead of selecting **Connect Entire Port to Bundle**, select **Disconnect Entire Port from Bundle**.

# Configuring Timeslot Assignment for Data Ports

The utilization of data port bandwidth is controlled by assigning timeslots, that is, bandwidth, to each subchannel:

- For an unframed data port, there is only one subchannel. The subchannel is configured as part of the data port physical layer parameters configuration procedure, as described in the *Specific Configuration Parameters for Unframed* Mode section on page *3-14*.
- For a framed data port, up to 10 subchannels can be defined (this capability is required in point-to-multipoint applications). Since each bundle serving a voice port of the corresponding VC-E1/16 is configured to connect to a specific subchannel (see *Configuring Bundle Routing Parameters* section on page *3-20*), this also assigns transmission bandwidth on the connection to the network for each bundle.

The data port timeslots used by a subchannel can be selected in two ways:

- Select individual timeslots on the corresponding VC-E1/16 data port. For the supervision terminal, use **Config›Connection›TS Assignment**.
- When a subchannel will use the full bandwidth of a whole VC-E1/16 data port, the connection of the whole port to the subchannel can be initiated either from **Config›Connection›TS Assignment** or from the E1 data port physical parameters configuration screen.

## Selecting Individual Timeslots

When you select a specific VC-E1/16 data port, you are provided with a timeslot map that includes timeslots 1 to 31.

➤ **To configure individual timeslot assignment:**

1. For each timeslot, you need to select the subchannel. Move the cursor under the timeslot in the **Sub Channel** row, and then select **Change Cell** and type the prescribed subchannel number, in the range of 1 to 10. **0** means that no subchannel is connected to the timeslot.
2. For each timeslot, you need to select its type: **CONNECTED** or **NC** (not connected to the selected subchannel). The selection is made on a list that appears under the table when the cursor is within the **TS Type** row. Move the cursor under the timeslot, and then select **Change Cell** to make the selection.

***Note***

*As an alternative, you can first use the **Assign Entire TSs to Port** to preconfigure the timeslots and their types, and then modify only the different timeslots.*

## Assigning all the Port Timeslots to a Subchannel

You can select all the timeslots of a VC-E1/16 data port to one subchannel in one step, using **Assign Entire TSs to Port** on the **Config›Connection›TS Assignment** screen, or on the corresponding **Config›Physical Layer›I/O›E1 Port Data** screen.

After this step, you can modify the configuration of any desired timeslots as explained in the *Selecting Individual Timeslots* section above.

➤ **To assign all the port timeslots:**

1. Select **Connect Entire Time Slots to Port** to display the configuration screen.
2. Select **Sub Channel Number** and enter the prescribed number.
3. Confirm the connection by selecting **Connect Entire Time Slots to Bundle.**

### Disconnecting a Subchannel from a Port

You can disconnect all the timeslots of a VC-E1/16 data port that are connected to a specific subchannel in one step, using the procedure described above but with one difference: in *Step 5*, instead of selecting **Connect Entire Time Slots to Port**, select **Disconnect Entire Time Slots from Port**.

## 3.10 Configuring Inband Management via VC-E1/16 Data Ports

VC-E1/16 installed in a Gmux-2000 support the transfer of inband management traffic via their data ports. Therefore, a remotely located network management station, for example, RADview, connected to a GbE port of one Gmux-2000 unit (referred to as a ***management gateway***) can manage other Gmux-2000 units, as well as Vmux-2100 Voice Trunking Gateways, connected to the management gateway via VC-E1/16 data ports.

***Note***

*Refer to the Gmux-2000 Installation and Operation Manual for a detailed description of management capabilities, and general management configuration instructions.*

The following procedure assumes that the Gmux-2000 serving as management gateway has already been configured for management by the desired management station via one of its GbE ports, in accordance with the Gmux-2000 Installation and Operation Manual. The procedure has two sections:

- Configuring the management gateway to support inband management of remote units via VC-E1/16 data ports
- Configuring each managed unit for management via VC-E1/16 data ports.

➤ **To configure the management gateway:**

1. For configuration by means of the supervision terminal, use **Configuration >Management >Managers List**.
2. Make sure that the management station is already configured in the **Managers List** as connected via the appropriate GbE port.
3. Add the desired VC-E1/16 module as a new manager:
   - Select **Add** on the **Managers List**.
   - Select **Interface**, and specify the desired VC-E1/16 module (identified by its slot number, for example, **VMUX SLOT 3**).

- Select **IP Address**, and enter the management address of the unit to be managed (for example, for a Gmux-2000 enter the IP address assigned to the **Online CL** on its **Host IP** screen).
- Select **Yes** for **Trap (for Manager**).
- Do not change the other defaults (leave **Next Hop** as **0.0.0.0**, and **VLAN Valid** as **NO**).

➤ **To configure a managed Gmux-2000 unit:**

1. For configuration by means of the supervision terminal, use **Configuration>Management>Managers List**.
2. Make sure the management IP address of the managed unit (the IP address assigned to the **Online CL** on the managed unit **Host IP** screen) is within the subnet of the management gateway IP address.
3. Select an appropriate VC-E1/16 module (one of the data ports of this module must be connected to a data port of the VC-E1/16 in the management gateway that has been configured as explained above).
4. Add the selected VC-E1/16 module as a new manager:
   - Select **Add** on the **Managers List**.
   - Select **Interface**, and specify the selected VC-E1/16 module (identified by its slot number, for example, **VMUX SLOT 1**).
   - Select **IP Address**, and enter the IP address of the management station (the same as for the management gateway).
   - Select **Next Hop**, and enter the management gateway IP address (appears next to **Online CL** on the management gateway **Host IP** screen).
   - Select **Yes** for **Trap (for Manager**).
   - Leave **VLAN Valid** as **NO**.

***Note*** *The same steps are needed for configuring a managed Vmux-2100 Voice Trunking Gateway. Refer to the Vmux-2100 Installation and Operation Manual for detailed instructions.*

## 3.11 Configuring N+1 Protection

VC-E1/16 and VC-E1/12 modules using the US_NS mode support the N+1 protection mode, one of the Gmux-2000 APS functions.

N+1 protection for VC-E1/16 or VC-E1/12 modules is configured as follows:

➤ **To configure N+1 protection for VC-E1/16 or VC-E1/12 modules:**

1. Open the **Config>System>Redundancy>I/O** screen and type **X** to add a new APS group.

***Note*** *Gmux-2000 supports a single N+1 protection group per chassis. Therefore, if another N+1 protection group is already configured, first delete it.*

2. Open the **APS Configuration** screen for the new APS group, and select **N+1** for the **Config Mode**. You may also add a name for the group, in the **Name** field.
3. Select the desired **Recovery Mode**:
   - **Non-Revertive** – after a protection flip, the protection module continues carrying the traffic even if the protected module replaced by it returns to normal operation.
   - **Revertive** – after a protection flip, the protection module returns to idle when the protected module replaced by it returns to normal operation, and thus is again capable of protecting the other modules. You can specify the **Time to Restore**, which ensures a minimum interval before flipping back.
4. Open the **APS Mapping** screen, and add the protection members in the following order (the order is indicated by the automatically assigned numbers in the **CH NUM** column):
   1. The first module to be specified is the VC-E1/16 or VC-E1/12 module serving as the protection module, in the format **{Slot]/{Port]**, where the port number must be **0**.

***Note***

- *After a protection module is selected, it cannot be changed. To select another module, first delete the N+1 protection group, and then reconfigure another group with the required protection module.*
- *It is not necessary to define configuration parameters for the protection module: the appropriate configuration parameters will be automatically downloaded by the CONTROL module when flipping is necessary.*

   2. After the protection module is specified, you can specify the protected VC-E1/16 or VC-E1/12 modules in any order. For each module, use the format **{Slot]/{Port]**, where the port number must be **0**.
      - For a Gmux-2000 equipped with a single STM1 module, the maximum number of VC-E1/16 or VC-E1/12 modules that can be protected is 5
      - For a Gmux-2000 equipped with two STM1 modules, the maximum number of VC-E1/16 or VC-E1/12 modules that can be protected is 4.
   3. To provide differential quality of service, select for each module its protection **Priority**: **Low** or **High**: in the event that protection is required for two modules at the same time, only the high priority module will be protected.

# Chapter 4

# Troubleshooting and Diagnostics

## 4.1 Overview

This Chapter explains the module-specific diagnostic functions of the VC-E1/16 module.

The VC-E1/16 diagnostic functions include:

- Sanity checks, detection of alarm conditions, and event collection. These functions are general Gmux-2000 diagnostic functions, and therefore the messages associated with VC-E1/16 modules are explained in *Chapter 7* of the *Gmux-2000 Installation and Operation Manual*.
- Collection of VC-E1/16 performance monitoring and status data – presented in *Section 4.2*.
- Diagnostic tests for checking transmission paths – presented in *Section 4.3*.
- Hardware troubleshooting instructions and solutions for frequently encountered problems – presented in *Sections 4.4* and *4.5*.

The information presented in this Chapter supplements the general Gmux-2000 diagnostics instructions contained in *Chapter 7* of the *Gmux-2000 Installation and Operation Manual*.

If you need additional support for this product, see *Section 4.6* for technical support information.

## 4.2 Performance Monitoring

The VC-E1/16 collects transmission performance and status data for its various ports and for the connections (bundles) terminated at the module ports.

The collected data enables the system administrator to monitor the transmission performance and thus the quality of service provided to users, for statistical purposes. In addition, when problems are reported by users served by VC-E1/16, the collected data can be used for diagnostic purposes, because it can help identify the source of the problem.

For TDM ports, the basic performance data is calculated for each second, and accumulated and displayed over a 15-minute (900 second) interval. The data accumulated for each interval, up to a maximum of 96 previous intervals (that is,

over the last 24-hour interval) is stored and can also be displayed. The oldest interval is 1, and the most recent one is the interval with the largest number. Only the data for the most recent 96 intervals is stored, and therefore the oldest data is overwritten by new data.

The performance statistics and status data is continuously collected, and is stored as long as the equipment operates. The performance statistics cannot be erased, but status data can be cleared by manual commands.

The stored data is deleted when the VC-E1/16 is reset or removed, and is also lost when the Gmux-2000 is powered down.

## Accessing the Performance Monitoring Functions

When using the Gmux-2000 supervision utility, the performance monitoring functions are accessed under the **Monitoring** menu, covered by *Chapter 7* of the *Gmux-2000 Installation and Operation Manual*.

The main monitoring activities include:

- Display performance monitoring statistics and status information for VC-E1/16 physical interfaces
- Display performance and status information for connections (bundles) terminated on VC-E1/16 ports.

## Displaying Physical Layer Performance Monitoring Data

The physical layer performance monitoring data includes:

- Statistics data. Physical layer statistics are available for both user-side (voice) ports and for network-side (data) ports. For each port, you can see:
  - Data for the current 15-minute interval
  - Data for any previous interval for which valid data is available. The maximum number of previous intervals is 96.
- Status data:
  - HDLC transport data for each voice (user side) and data (network side) port.
  - Voice transport data for each timeslot of each voice (user) port.
  - Signaling status for each timeslot of each voice (user) port. The timeslot signaling data can be separately displayed for each direction (receive and transmit).
  - Bandwidth control data for each voice port.

### Displaying VC-E1/16 Physical Layer Statistics

The physical layer statistics enables detecting transmission errors on the links connected to the various ports.

The statistics data reported by VC-E1/16 modules (see *Table 4-1*) enable analyzing the transmission performance of the various ports, to detect

anomalous conditions, for example, many unavailable seconds (UAS) that may impair service:

- When everything operates normally, the transmission performance of E1 links meeting the internationally-accepted standards is very good, resulting in small numbers of coding errors (CVs and LCVs) and few errored seconds (ES).
- Small numbers of randomly distributed errors have little effect on voice quality, but error bursts may seriously degrade the quality of compressed voice, and their sources must be found and corrected in order to restore satisfactory performance. Usually, error bursts are reflected in the number of severely errored seconds (SES).

➤ **To display physical layer statistics data:**

1. To display statistics data, select the **Statistics** monitoring task.
2. Select the **Physical Layer** statistics option.
3. Select the desired VC-E1/16 module by specifying its I/O slot number.
4. Select the desired type of ports.

   You can select only a type of ports available on the module installed in the selected I/O slot. The port types depend on the VC-E1/16 operating mode, as listed in *Table 4-1*.

*Table 4-1. VC-E1/16 Port Types versus Operating Mode*

| Card Operating Mode | User Ports Type | Network-Side Ports Type |
|---|---|---|
| **VC16E1_UE_NE** | E1 voice ports (1 to 12 for VC-E1/12; 1 to 16 for VC-E1/16) | E1 data ports (1 or 2) |
| **VC16E1_UE_NS** | E1 voice ports (same as above) | VC-12 ports (17 to 21) |
| **VC16E1_US_NE** | VC-12 ports (same as above) | E1 data ports (1 or 2) |
| **VC16E1_US_NS** | VC-12 ports (same as above) | VC-12 ports (17 to 21) |

5. Select the type of interval: current interval, or a previous interval.

***Note***

*When you select the current interval, you can also see the current number of valid intervals. If the number of valid intervals is less than 96, it will increase by 1 when the current interval expires and a new interval is opened: however, you will see the change only after refreshing the display.*

6. Select a specific port. The range of ports depends on the VC-E1/16 model and type of ports, as listed in *Table 4-1*.
7. After selecting a specific port, you will see the requested statistics data. The displayed parameters are explained in *Table 4-2*.

When you display data on previous intervals, you can see the port data for other valid intervals by entering a different interval number, without having to go again through the whole selection process.

*Table 4-2. Physical Layer Statistics Parameters*

| Parameter | Description |
|---|---|
| **Interval Number** | The number of the 15-minute interval to be displayed, in the range of 1 to 96.<br>This parameter is available only for previous intervals.<br>The default value is 1. However, you can also select another interval, up to the number displayed in the **Valid Intervals** field. |
| **Valid Intervals** | The number of previous intervals (up 96) for which performance monitoring information can be displayed.<br>This parameter is available only for the current interval |
| **Elapsed Time** | The elapsed time (in seconds) since the beginning of the current interval, in seconds. The range is 1 to 900 seconds.<br>This parameter is available only for the current interval |
| **ES** | Displays the number of errored seconds (ESs) in the selected interval.<br>An errored second is a second with one or more coding violations (CVs or LCVs), and/or one or more of the following defects: reception of AIS, loss of line signal (for external ports only). |
| **SES** | Displays the number of severely errored seconds (SESs) in the selected interval.<br>A SES is any second with more than 40 coding violations (CVs or LCVs), and/or one or more of the defects taken into consideration for an ES. |
| **UAS** | Displays the number of unavailable seconds (UASs) in the selected interval.<br>UAS is declared after 10 continuous SESs. During UASs, all the other defect counters are freezed. |
| **CV** | Displays the number of coding violations (CV) in the selected interval.<br>A coding violation is declared when a Bit Interleaved Parity (BIP) error is detected in the incoming signal, and it indicates that errors have been detected in the received signal.<br>This parameter is available only for internal (VC-12) ports |
| **LCV** | Displays the number of line coding violations (LCV) in the selected interval.<br>A line coding violation is declared when a line signal decoding error occurs at the receive side (such errors commonly occur when the receive signal level is too low, usually as a result of excessive line attenuation between the transmit and receive ports).<br>This parameter is available only for external (E1) ports |

## Displaying VC-E1/16 Physical Layer HDLC Transport Status Data

HDLC transport status monitoring data are available for both voice (user side) and data (network side) ports.

- For voice ports, the HDLC protocol is used to transport CCS data. As explained in Chapter 3, for each voice port it is possible to assign up to two timeslots for CCS transport:
  - One timeslot (referred to as **HDLC Data Flow A**) can be assigned to transparent transport of HDLC frames. This timeslot can also transport other protocols, including Signaling Scheme #7 (SS7).
  - A second timeslot (referred to as **SS7 Data Flow A**) can be assigned to optimized transport of Signaling Scheme #7 protocol data units (PDUs). This timeslot should not be used for other protocols, because it is not transparent.
- For data ports, the HDLC protocol is used for end-to-end handshaking and management traffic. One HDLC channel is used on each data port subchannel. When using a framed mode, multiple subchannels (1 to 10) can be configured, and therefore it is necessary to select a specific subchannel for display. However, when a data port uses the unframed mode (whole port serves as a single subchannel), only one HDLC channel is available and no selection is needed.

The HDLC transport data can be analyzed to troubleshoot unsatisfactory performance, or compressed voice transmission failures.

➤ **To display HDLC transport status data for voice ports:**

1. Select the **Status** monitoring task.
2. Select the **Physical Layer** option.
3. Select the desired VC-E1/16 module by specifying its I/O slot number.
4. Select the E1 port type – voice (see *Table 4-1*).
5. Select the **HDLC Counters** option.
6. Select the desired voice port (see *Table 4-1*).
7. Select the desired HDLC channel: **SS7 Data Flow A** or **HDLC Data Flow A**.
8. When ready, send a **Display** command.

***Note***

*You can reset the status counters by a C (clear) command.*

*Table 4-3* presents the HDLC transport status monitoring parameters for voice ports.

***Note***

- ***Rx*** *– identifies the receive direction (from the local user's equipment to the VC-E1/16 port)*
- ***Tx*** *– identifies the transmit direction (from the VC-E1/16 port to the local user's equipment)*
- *Unless a specific time interval is listed, the displayed values represent the total accumulated on the corresponding HDLC channel since the equipment has been last turned on, or since the last module reset or counters clear command (whichever occurred last).*

*Table 4-3. HDLC Transport Status Monitoring Parameters for Voice Ports*

| Parameter | Description |
|---|---|
| **Rx Total Frames** | Total number of received protocol packets |
| **Rx Total Bytes** | Total number of received protocol data octets |
| **Rx 60 sec Frames** | Number of protocol packets received during the last 60 seconds |
| **Rx 60 sec Bytes** | Number of protocol data octets received during the last 60 seconds |
| **RX LG Err** | Total number of received protocol packets that had length violations |
| **Rx NO Err** | Total number of received protocol packets which were not octet-aligned |
| **Rx CR Err** | Total number of protocol packets received with CRC errors |
| **Rx AB Err** | Total number of received protocol packets with abort sequence violations (7 or more consecutive ones) |
| **Rx OV Err** | Total number of receive buffer overrun events |
| **Rx Busy Count** | Total number of times the received protocol packets were not processed because no resources were available |
| **Q50 Local State** | The Q.50 bandwidth control status at the local port: **Enable** new calls, or **Disable** (reject) new calls |
| **Q50 Remote State** | The Q.50 bandwidth control status at the far end equipment connected to the local port: **Enable** new calls, or **Disable** (reject) new calls |
| **Tx Total Frames** | Total number of protocol packets transmitted |
| **Tx Total Bytes** | Total number of protocol data octets transmitted |
| **Tx 60 sec Frames** | Number of protocol packets transmitted during the last 60 seconds |
| **Tx 60 sec Bytes** | Number of protocol data octets transmitted during the last 60 seconds |
| **Tx UN Err** | Total number of transmit buffer underrun events |
| **Tx Over Discard** | Total number of protocol packets intended for transmission that have been discarded due to lack of buffer resources |
| **Tx Q Max Frames** | The maximum number of protocol packets that were queued at any time in the transmit queue |
| **Tx Q Cur Frames** | The current number of protocol packets that are waiting for transmission in the transmit queue |

➤ **To display HDLC transport status data for data ports:**

1. Select the **Status** monitoring task.
2. Select the **Physical Layer** statistics option.
3. Select the desired VC-E1/16 module by specifying its I/O slot number.
4. Select the E1 port type – data (see *Table 4-1*).
5. Select the **HDLC Counters** option.
6. Select the desired data port (see *Table 4-1*).
7. Select the desired subchannel (relevant only when using a framed mode), in the range of 1 to 10.
8. When ready, send a **Display** command.

***Note***

*You can reset the status counters by a C (clear) command.*

*Table 4-4* presents the HDLC transport status monitoring parameters for data ports.

***Note***

- ***Rx** – identifies the receive direction (from the network to the VC-E1/16 port)*
- ***Tx** – identifies the transmit direction (from the VC-E1/16 port to the network)*
- *Unless a specific time interval is listed, the displayed values represent the total accumulated on the HDLC channel since the equipment has been last turned on, or since the last module reset or counters clear command (whichever occurred last).*

*Table 4-4. HDLC Transport Status Monitoring Parameters for Data Ports*

| Parameter | Description |
|---|---|
| **Rx Total Frames** | Total number of received protocol packets |
| **Rx Total Bytes** | Total number of received protocol data octets |
| **Rx 60 sec Frames** | Number of protocol packets received during the last 60 seconds |
| **Rx 60 sec Bytes** | Number of protocol data octets received during the last 60 seconds |
| **RX LG Err** | Total number of received protocol packets that had length violations |
| **Rx NO Err** | Total number of received protocol packets which were not octet-aligned |
| **Rx CR Err** | Total number of protocol packets received with CRC errors |
| **Rx AB Err** | Total number of received protocol packets with abort sequence violations (7 or more consecutive ones) |
| **Rx OV Err** | Total number of receive buffer overrun events |
| **Rx Busy Count** | Total number of times the received protocol packets were not processed because no resources were available |
| **Tx Total Frames** | Total number of protocol packets transmitted |
| **Tx Total Bytes** | Total number of protocol data octets transmitted |
| **Tx 60 sec Frames** | Number of protocol packets transmitted during the last 60 seconds |

| Parameter | Description |
|---|---|
| **Tx 60 sec Bytes** | Number of protocol data octets transmitted during the last 60 seconds |
| **Tx UN Err** | Total number of transmit buffer underrun events |
| **Tx Over Discard** | Total number of protocol packets intended for transmission that have been discarded due to lack of buffer resources |
| **Tx Q Max Frames** | The maximum number of protocol packets that were queued at any time in the transmit queue |
| **Tx Q Cur Frames** | The current number of protocol packets that are waiting for transmission in the transmit queue |
| **Aver Outgoing Bw** | The average data rate, in bps, measured for the transmit direction of the selected subchannel |
| **Max Outgoing Bw** | The peak data rate, in bps, that has been measured for the transmit direction of the selected subchannel |
| **Configured Bw** | The user-configured bandwidth, in bps, for the selected subchannel (this number is determined by the number of timeslots assigned to the selected subchannel: for example, 10 timeslots result in a bandwidth of 640000 bps).<br>Comparing the three bandwidth |
| **Q50 Local State** | The Q.50 bandwidth control status at the local port: **Enable** new calls, or **Disable** (reject) new calls |
| **Q50 Remote State** | The Q.50 bandwidth control status at the far end equipment connected to the local port: **Enable** new calls, or **Disable** (reject) new calls |
| **Modem VBD State** | The current state of the call restriction function for voiceband modem calls: **Enable** calls, or **Disable** (block) calls, because the current number of active voiceband modem calls equals the maximum user-configured number of voiceband modem calls |
| **Modem Relay State** | The current state of the call restriction function for modem calls using relaying: **Enable** calls, or **Disable** (block) calls, because the current number of active relay-mode modem calls using relaying equals the maximum user-configured number of relay-mode modem calls |

## Displaying VC-E1/16 Physical Layer Voice Timeslot Status Data

The voice timeslot status monitoring function provides detailed information on the traffic carried by any voice timeslot transported by the VC-E1/16 (including traffic statistics and timeslot utilization data), and also data regarding the status of the internal VC-E1/16 digital signal processor (DSP) that provides voice compression services for the selected timeslot.

This capability provides a powerful tool for analyzing DSP condition, voice channel service quality, and for monitoring in real time the timeslot traffic. By resetting the status data at the desired instant, it is possible to ensure that only current, valid data is taken into consideration.

➤ **To display voice timeslot status:**

1. Select the **Status** monitoring task.
2. Select the **Physical Layer** statistics option.
3. Select the desired VC-E1/16 module by specifying its I/O slot number.
4. Select the E1 port type – voice (see *Table 4-1*).
5. Select the **Voice Counters** option.
6. Select the desired voice port (see *Table 4-1*).
7. Select the desired timeslot, in the range of 1 to 31. Skip timeslot 16 when using the G.704 multiframing mode, because in this case timeslot 16 carries signaling information (signaling information can be analyzed using the voice signaling status monitoring functions, explained on page *4-11*).
8. When ready, send a **Display** command.

***Note***

*You can reset the status counters by a C (clear) command.*

*Table 4-5* presents the voice timeslot status monitoring parameters.

***Note***

- ***Rx** – identifies the receive direction (from the VC-E1/16 port to the local user's equipment)*
- ***Tx** – identifies the transmit direction (from the local user's equipment to the VC-E1/16 port)*
- *The displayed values represent the total accumulated since the equipment has been last turned on, or since the last module reset or counters clear command (whichever occurred last).*

*Table 4-5. Voice Timeslot Status Monitoring Parameters*

| Parameter | Description |
|---|---|
| **Rx Total Voice** | Total number of compressed voice packets received from the DSP |
| **Tx Total Voice** | Total number of compressed voice packets transmitted by the DSP to the network |
| **Rx Total Silence** | Total number of silence packets received from the DSP |
| **Tx Total Silence** | Total number of silence packets transmitted by the DSP to the network |
| **Rx Total Fax** | Total number of fax packets received from the DSP |
| **Tx Total Fax** | Total number of fax packets transmitted by the DSP to the network |
| **Rx Total Modem** | Total number of modem packets received from the DSP |
| **Tx Total Modem** | Total number of modem packets transmitted by the DSP to the network |
| **Rx Total Dtm** | Total number of DTMF signaling packets received from the DSP |
| **Tx Total Dtm** | Total number of DTMF signaling packets transmitted by the DSP to the network |
| **Rx Total Mf** | Total number of MF signaling packets received from the DSP |
| **Tx Total Mf** | Total number of MF signaling packets transmitted by the DSP to the network |

| Parameter | Description |
|---|---|
| **Rx Silence Rate** | Average percentage of silence measured for the timeslot receive direction |
| **Tx Silence Rate** | Average percentage of silence measured for the timeslot transmit direction |
| **Total Underrun** | Total number of packets lost because of jitter buffer underrun events |
| **Total Overrun** | Total number of packets lost because of jitter buffer overflow events |
| **Total Heartbeat** | DSP heartbeat indication. During normal operation, the heartbeat count must increase continuously, at a rate of a counts per second).<br>If the displayed value is not steadily increasing, this may indicate a DSP error state: to restore normal operation, it is necessary to reset the VC-E1/16. |
| **Illegal Voice Packets** | Total number of illegal received voice packets |
| **Q50 64K Seize** | Total number of Q.50 64 kbps clear channel call set up requests received from the local user's equipment, for example, a PBX. Relevant only when the Q.50 mode is enabled |
| **Q50 Speech Seize** | Total number of Q.50 3.1 kbps voice channel call set up requests received from the local user's equipment. Relevant only when the Q.50 mode is enabled |
| **Q50 Release** | Total number of calls that were released by the VC-E1/16 port using the Q.50 protocol, after being established. Relevant only when the Q.50 mode is enabled |
| **Q50 Not Avail Co** | Total number of times the channel was reported as not available using the Q.50 protocol. Relevant only when the Q.50 mode is enabled |
| **Q50 Speech Pack** | Total number of Q.50 3.1 kbps voice channel call set up requests that have been accepted by the VC-E1/16 port. Relevant only when the Q.50 mode is enabled |
| **Q50 Speech Nack** | Total number of Q.50 3.1 kbps voice channel call set up requests that have been rejected by the VC-E1/16 port. Relevant only when the Q.50 mode is enabled |
| **Q50 64K Nack** | Total number of Q.50 64 kbps clear channel call set up requests that have been rejected by the VC-E1/16. Relevant only when the Q.50 mode is enabled |
| **Q50 Ts State** | Displays the current state of the Q.50 protocol:<br>• **OOS** – Q.50 is enabled, but signaling between the VC-E1/16 port and the local user's equipment is out of service<br>• **Used** – Q.50 is enabled and signaling between the VC-E1/16 port and the local user's equipment is active<br>• **Not Used** – Q.50 is not available on this channel |
| **Ts Mode** | Displays the current operational state of the selected timeslot:<br>• **Config** – the timeslot is under configuration<br>• **Idle** – the timeslot is idle<br>• **Setup** – a call is being setup on this timeslot<br>• **Voice** – the timeslot carries voice traffic<br>• **Data** – the timeslot carries modem traffic<br>• **Fax** – the timeslot carries fax traffic<br>• **VBD** – the timeslot carries voiceband modem traffic<br>• **Tandem** – the timeslot operates in the super-tandem mode<br>• **HDLC** – the timeslot carries HDLC data<br>• **Transp** – the timeslot carries data in the transparent mode (reserved)<br>• **Unknown** – the timeslot state is not known, and cannot be retrieved |

## Displaying VC-E1/16 Physical Layer Voice Port Signaling Status

VC-E1/16 enables displaying in real-time the signaling information for each payload timeslot (timeslots 1 to 15, and 17 to 31) of voice (user) ports using channel-associated signaling (that is, a G.704 multiframe mode), separately for the receive and transmit directions.

This capability provides a powerful tool for troubleshooting voice channel problems, because it enables monitoring in real time the timeslot signaling information.

The signaling information consists of four bits (designated ABCD in the applicable standards – see description in Appendix C of the Gmux-2000 Installation and Operation Manual). VC-E1/16 does not use the signaling information, but only forwards the information end-to-end. Generally, the information is transparently transferred, unless the user defines signaling profiles for translating the signaling information (see Chapter 3 for a description of signaling profiles).

Therefore, to interpret the displayed signaling information, make sure to obtain the list of signaling codes used by the user's equipment.

➤ **To display the voice timeslot signaling information:**

1. Select the **Status** monitoring task.
2. Select the **Physical Layer** statistics option.
3. Select the desired VC-E1/16 module by specifying its I/O slot number.
4. Select the E1 port type – voice (see *Table 4-1*).
5. Select the desired option: **Voice Rx Signaling Counters**, or **Voice Tx Signaling Counters**.

***Note***

- ***Rx*** *– identifies the receive direction (from the VC-E1/16 port to the local user's equipment)*
- ***Tx****– identifies the transmit direction (from the local user's equipment to the VC-E1/16 port).*

6. Select the desired voice port (see *Table 4-1*).
7. When ready, send a **Display** command.
8. You will a table with the timeslot signaling information.

   The signaling information consists of four bits (designated ABCD in the applicable standards – see Appendix C of the Gmux-2000 Installation and Operation Manual). The interpretation of these bits depends on the specific channel type.

## Displaying VC-E1/16 Physical Layer Bandwidth Control Status Data

VC-E1/16 supports a proprietary bandwidth control method that supplements the ITU-T Rec. Q.50 capabilities.

You can read the results of the bandwidth control function for each voice port, to evaluate the effects of bandwidth control. By resetting the status data at the

desired instant, it is possible to ensure that only current, valid data is taken into consideration.

➤ **To display the bandwidth control status:**

1. Select the **Status** monitoring task.
2. Select the **Physical Layer** option.
3. Select the desired VC-E1/16 module by specifying its I/O slot number.
4. Select the E1 port type – voice (see *Table 4-1*).
5. Select the **Bandwidth Control** option.
6. Select the desired voice port (see *Table 4-1*).
7. When ready, send a **Display** command.

***Note***

*You can reset the status counters by a C (clear) command.*

*Table 4-6* presents the bandwidth control status parameters.

*Table 4-6. Bandwidth Control Status Parameters*

| Parameter | Description |
|---|---|
| **Bw Ctrl Current State** | Displays the current bandwidth control state:<br>• **Normal** – normal operation mode, new calls are accepted<br>• **Discard** – calls can still be set up, but the traffic load is high and therefore the bandwidth control mechanism randomly discards voice paclets to prevent overload.<br>• **Disable New Calls** – set up requests for new calls are rejected because of excessive traffic load. |
| **Stop New Calls (Sec)** | Total number of seconds during which new calls have been blocked |
| **Discarded Voice Packets (Sec)** | Total number of seconds during which voice packets have been randomly discarded |
| **Max Delta Bw (Byte)** | Total packet traffic load reduction, in bytes, as a result of the bandwidth control mechanism.<br>If you want to reduce the effects of the bandwidth control mechanism, it is necessary to increase the available transport bandwidth. The increase should be sufficient to compensate for the average data rate reduction. You can estimate the long-term average data rate reduction as follows:<br>**Data Rate (bps) = 8 × Max Delta Bw (Byte)/Discarded Voice Packets (Sec)** |

## Displaying Connection Performance Monitoring Data

The connection (bundle) performance monitoring data displays the bundle state, identifies the exit port, and provides detailed information on the traffic carried by the bundle. See also the *Interpreting Sequence Numbers Data* section below.

This capability provides a powerful tool for analyzing bundle utilization, and for monitoring in real time the bundle traffic. By resetting the status data at the

desired instant, it is possible to ensure that only current, valid data is taken into consideration.

➤ **To display the connection (bundle) status:**

1. Select the **Status** monitoring task.
2. Select the **Connection** option.
3. Select the **Bundle Counters** option.
4. Select the desired **Bundle ID** by specifying its number, in the range of 1 to 2000. You can select only existing bundles.
5. After selecting a specific bundle, the bundle parameters are immediately displayed. The display includes the following information:
   - Bundle number and its type (TDMoIP (CV))
   - Bundle state:
     - **Connected** – the bundle is connected and can carry traffic
     - **Disconnected** – the bundle is administratively disconnected (it is configured with **Connect** = **NO**)
     - **IP OOS** – the bundle cannot carry traffic, because the IP link to its destination is out-of-service. To correct the problem, first check that the exit port and the IP communication parameters are correctly configured (that is, correct destination IP address, and correct next hop IP address). If IP configuration is correct, the problem may be in the network: you may use the ***ping*** function to check.
     - **Echo Wait** – the IP link is up, but the bundle cannot carry traffic because Layer 4 (UDP) is down. In this case, first check the source and destination CBID parameters.
   - The bundle exit port, for example, the identification of the TDM link (A, B, C, D) or the GbE module and port
   - Bundle counters section, which displays the bundle performance monitoring parameters.

***Note***

*You can reset the bundle counters by a C (clear) command.*

*Table 4-7* presents the bundle performance monitoring parameters.

***Note***

- ***Rx*** *– identifies the receive direction (from the network to the VC-E1/16)*
- ***Tx*** *– identifies the transmit direction (from the local VC-E1/16 to the network)*
- *Unless a specific time interval is listed, the displayed values represent the total accumulated since the bundle has last been connected, or since the last counters clear command (whichever occurred last).*

*Table 4-7. Bundle Performance Monitoring Parameters*

| Parameter | Description |
|---|---|
| Rx Total Frames | Total number of packets received |
| Tx Total Frames | Total number of data packets transmitted |
| Rx 60 sec Frames | Number of packets received during the last 60 seconds |
| Tx 60 sec Frames | Number of packets transmitted during the last 60 seconds |
| Rx Total Bytes | Total number of received data octets |
| Tx Total Bytes | Total number of transmitted data octets |
| Rx 60 sec Bytes | Number of data octets received during the last 60 seconds |
| Tx 60 sec Bytes | Number of data octets transmitted during the last 60 seconds |
| Tx on Timeout | Total number of packets transmitted before reaching the maximum configured size, as a result of reaching the maximum configured packetizing interval |
| Tx 60 sec Timeout | Number of packets transmitted before reaching the configured size, as a result of reaching the maximum configured packetizing interval during the last 60 seconds |
| Tx Max Size | Total number of packets transmitted as a result of reaching the maximum configured size |
| Tx 60 sec Max Size | Number of packets transmitted as a result of reaching the maximum frame configured size during the last 60 seconds |
| Rx Total Sg | Total number of received signaling packets |
| Tx Total Sg | Total number of transmitted signaling packets |
| Rx Total Voice | Total number of received voice packets |
| Tx Total Voice | Total number of transmitted voice packets |
| Rx 60 sec Voice | Number of voice packets received during last 60 seconds |
| Tx 60 sec Voice | Number of voice packets transmitted during last 60 seconds |
| Rx Total Hdlc | Total number of received data packets |
| Tx Total Hdlc | Total number of transmitted HDLC packets |
| Rx 60 sec Hdlc | Number of HDLC packets received during last 60 seconds |
| Tx 60 sec Hdlc | Number of HDLC packets transmitted during last 60 seconds |
| Rx Silence Rate | Average percentage of silence measured for the receive direction on all the timeslots connected to the bundle |
| Tx Silence Rate | Average percentage of silence measured for the transmit direction on all the timeslots connected to the bundle |
| Freed Seq Num | Total number of duplicate packets that were received and dropped (the duplication occurs within the network) |
| Miss Seq Num | Total number of packets that did not arrive in time, as expected according to their sequence number (either because they were lost in the network, or reordered) |
| Longest Miss Seq | The largest interval of consecutively numbered missed packets |

## Interpreting Sequence Numbers Data

In accordance with the applicable standards, transmitted packets carry a sequence number that is automatically assigned. The packet sequence number is a 16-bit field (resulting range is 0 to 65535). Consecutive packets are automatically consecutive sequence numbers. When the sequence number reaches the maximum value (65535), it is reset to 0.

Sequence number statistic data can be used to trace problems in the network (IP, PDH or SDH) connecting the VC-E1/16 to other devices. For this purpose, the VC-E1/16 module analyzes the sequence numbers of the packets received from the network. The results of this analysis are presented by means of three parameters:

- **Miss Seq Num** parameter: displays the total number of missing packets. Only the total number of missing packets is relevant, not which specific packets are missing. If the network would have 100% reliability, all the transmitted packets would reach their destination, and this parameter would be 0.

  For example, if the received packets are numbered 1, 2, 3, 5, 6, 7, 8, then packet 4 is missing, and the missing packet counter is incremented by 1.

- **Longest Miss Seq**: displays the largest sequence of consecutive missing packets. When this parameter reaches a large value, it indicates that the network drops groups of packets in bursts.

  For example, if the received packets are numbered 39, 41, 42, 43, 44, 45, 46, ...., the longest burst is 4 (because 4 consecutive packets, numbered 37, 38, 39, 40, are missing).

- **Freed Seq Num:** displays the number of duplicated packets that were received. Packets with duplicate sequence numbers are dropped by the VC-E1/16 receive path, and therefore a large number of dropped, or freed, packets indicates a duplication problem in the network (in packet networks, this often happens when packets can reach their destination over more than one route).

  For example, if the received packets are numbered 6, 7, 8, 9, 6, 6, 10, 11, ...., then packet 6 was duplicated in the network (appeared a total of three times at the receive side). The two duplicated packets are dropped by the VC-E1/16, and the freed sequence number counter is incremented by 2.

The tracking window used to detect missing or duplicated packets is 30 packets wide. Missing and duplicated packets will be recognized only if their sequence numbers are within this tracking window.

For example, if the received packets are numbered 96, 97, 98, 99, 100, 75, ..., then packet 75 is recognized as a duplicate packet (100 – 75 = 25, which is less than 30, and therefore the packet is inside the window). Therefore, packet 75 is **freed,** and therefore VC-E1/16 now expects packet 101 to arrive from the network. However, if the received packets are numbered 96, 97, 98, 99, 100, 65, ..., then packet 65 cannot be recognized as a duplicate and is **accepted** (100 – 65 = 35, which is larger than 30, and therefore it is outside the window). Now the card expects packet 66 to arrive: if packet 101 would arrive instead, then the **Longest Miss Seq** parameter will indicate 35.

***Note*** *Bundle statistics must be cleared after the bundle has gone up, otherwise the sequence number mechanism is irrelevant. This is because the sequence number fields are meaningless during the bundle set-up time (that is, during the time it takes the bundle to go into the* ***connected*** *state).*

## 4.3 VC-E1/16 Diagnostic Functions

In addition to the power-up self-test, the VC-E1/16 module also supports the following diagnostic functions:

- Voice ports: local and remote loopbacks, and test tone injection toward the local equipment
- Data ports: remote loopback.

The diagnostic functions are controlled by means of the Diagnostics>Loopback menu. This section describes the VC-E1/16 module loopbacks.

### Voice Port Diagnostic Functions

#### Remote Loopback on Voice Port

The remote voice port loopback returns the E1 receive signal of the tested port toward the local equipment connected to the port.

The loopback is activated at the line side of the E1 framer of the tested port, which receives the signal through the SDH mapper (for ports connected to the SDH network), or the line interface unit (LIU) of the tested port.

- When a remote loopback is activated on an external voice port, the E1 framer of that port returns the receive signal towards the LIU, via the transmit path of the same port. *Figure 4-1* shows the signal paths when a remote loopback is activated on an external voice port.

  This test checks the connections to the local voice port, including the transmission plant connecting the local user's equipment to the VC-E1/16 voice port, and the E1 line interface of the VC-E1/16 voice port.

- When the loopback is activated on a voice port connected to the SDH network (that is, connected to an SDH port located on the STM1 module via the Gmux-2000 telecom buses), the E1 framer returns the received signal to the SDH mapper, via the transmit path of the same port. *Figure 4-2* shows the signal paths when a remote loopback is activated on a voice port connected to the SDH network.

  This test checks the transmission path to the local voice port, including the SDH transmission plant, the STM1 module, the internal Gmux-2000 paths connecting the user's equipment to the local VC-E1/16 voice port, and the E1 framer of the VC-E1/16 voice port.

In both configurations, the test signal is provided by the equipment connected to the local voice port, that must receive its own transmission. Note that as long as

the test is activated, that equipment cannot use loopback timing: it must use either internal timing, or lock its timing to an external clock signal.

While the loopback is activated, the signal received from the remote voice port is ignored. The signal received from the local user's equipment remains connected to the DSP, which continues sending the received payload for transmission through the network (E1 or SDH) to the equipment at the remote end of the link.

To ensure that the user's equipment connected to the tested port is capable of providing a good signal, the remote voice port loopback should be activated on the local VC-E1/16 port only after checking that the end user's equipment operates normally while its own local loopback is activated.

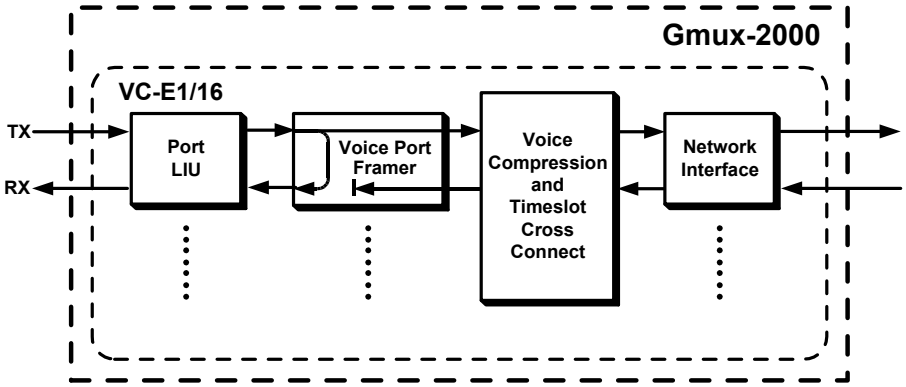



*Figure 4-1. Remote Loopback on External Voice Port*

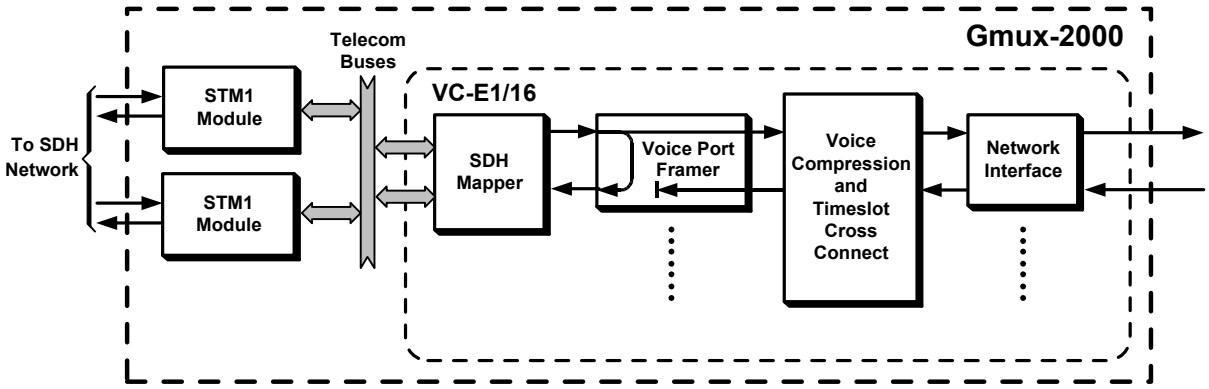



*Figure 4-2. Remote Loopback on Voice Port Connected to SDH Network*

## Local Loopback on Voice Port

The local voice port loopback returns the E1 signal restored from the compressed voice stream received from the remote end (through the E1 or SDH network connection), toward the equipment at the remote end of the link.

The loopback is activated at the digital side of the E1 framer of the tested port. The E1 framer returns the signal restored by the receive path of the DSP serving the port towards the DSP, via the transmit path of the same port.

***Note***

*When the tested port receives multiple bundles, each bundle's signal is independently looped back.*

- *Figure 4-3* shows the signal paths when a local loopback is activated on an external voice port. This loopback fully checks the operation of the local VC-E1/16 voice port, except for the E1 line interface (LIU); it also checks the internal VC-E1/16 signal paths that end at the tested VC-E1/16 port, including the transmission path through the network connecting the remote equipment to the local Gmux-2000, and the connections of the remote equipment to the far end Gmux-2000.
- When the local loopback is activated on a voice port connected to the SDH network (that is, connected to an SDH port via the Gmux-2000 telecom buses), the tested signal paths do not change. As shown in *Figure 4-4*, the only difference is in the paths not included with the loopback.

  This test checks the transmission path to the local voice port, including the SDH transmission plant, the STM1 module, the internal Gmux-2000 paths connecting the user's equipment to the local VC-E1/16 voice port, and the E1 framer of the VC-E1/16 voice port.

In both configurations, the test signal is provided by the equipment connected to the remote end, that must receive its own transmission. Note that as long as the test is activated, that equipment cannot use loopback timing: it must use either internal timing, or lock its timing to an external clock signal.

While the local loopback is activated, the transmit signal arriving from the local user's equipment is ignored, but the local voice port continues sending the received signal to the local equipment.

To ensure that the remote equipment is capable of providing a good signal, the local loopback should be activated on the local VC-E1/16 voice port only after checking that the remote end user's equipment connected to the tested port operates normally while its own local loopback is activated.

With the local loopback is activated on the local VC-E1/16 port, the remote end user's equipment must receive its own signal, and thus it must be frame-synchronized.

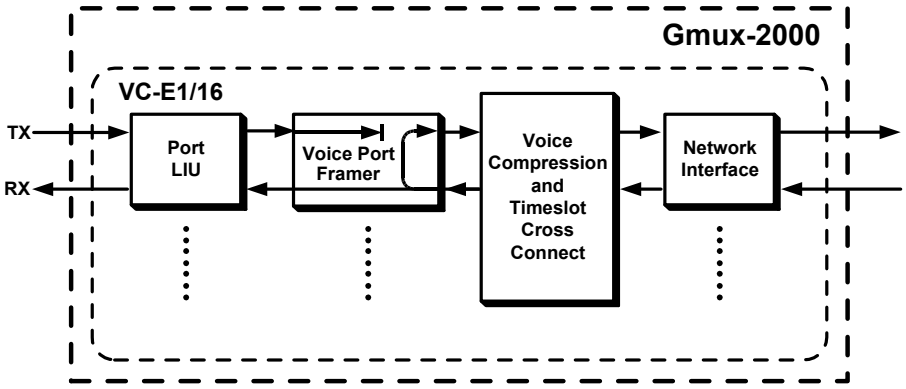



*Figure 4-3. Local Loopback on External Voice Port*

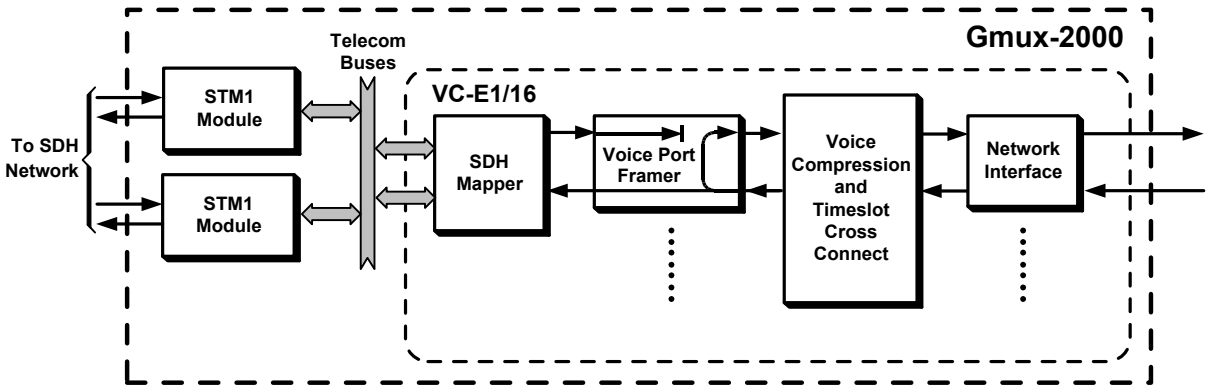



*Figure 4-4. Local Loopback on Voice Port Connected to SDH Network*

## Test Tone Injection

The test tone is a data sequence repeating at a rate of approximately 1 kHz. This data sequence is identical to the data sequence that would have been generated if a signal at the test frequency, having a nominal level of 1 mW (0 dBm0), were applied to the input of the DSP serving the voice port.

*Figure 4-5* shows the signal paths.

The test tone can be injected in any desired timeslot of a voice port, toward any desired local voice channel. When the local test tone injection is activated, the data received from the remote end in the corresponding timeslot of the local voice port is replaced by the test tone sequence. The resulting analog signal is therefore supplied to the local subscriber. As a result, a 1-kHz tone should be heard clearly and loudly in the earpiece of a telephone set connected to the corresponding voice channel. Test equipment can also be used to measure the test tone level; in this case, note that the nominal test tone level is changed in accordance with the **Volume to Line** parameter (see Configuring Voice Parameters section in Chapter 3) selected for the tested timeslot.

The signal transmitted by the voice channel toward the other end remains connected to the corresponding transmit timeslot.

At any given time, test tone injection can be activated in timeslots of only one voice port of the VC-E1/16 module. The number of timeslots is not limited.

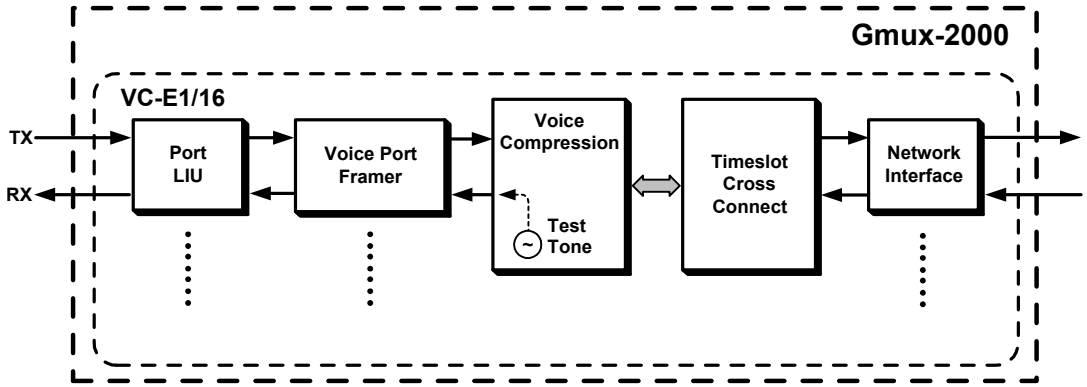



*Figure 4-5. Test Tone Injection Path*

# Data Port Diagnostic Functions

## Remote Loopback on Data Port

The remote data port loopback returns the E1 receive signal of the tested port toward the network, via the network connection terminated at the tested port.

The loopback is activated at the line side of the E1 framer of the tested port, which receives the signal through the SDH mapper (for ports connected to the SDH network), or the line interface unit (LIU) of the tested port.

- When a remote loopback is activated on an external data port, the E1 framer of that port returns the receive signal towards the LIU, via the transmit path of the same port. *Figure 4-6* shows the signal paths when a remote loopback is activated on an external data port.

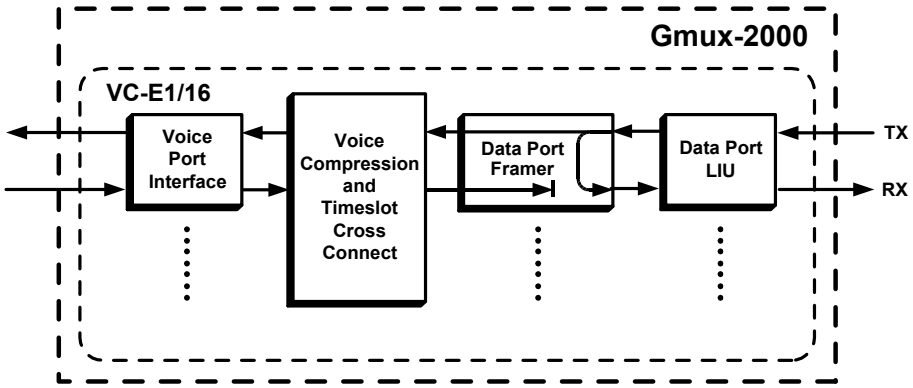



*Figure 4-6. Remote Loopback on External Data Port*

This test checks the whole transmission path between any remote user's equipment that connects through the network to the local data port, including the interconnecting transmission and switching plant; it also tests the LIU of the VC-E1/16 data port.

- When the loopback is activated on a data port connected to the SDH network (that is, connected to an SDH port via the Gmux-2000 telecom buses), the E1 framer returns the received signal to the SDH mapper, via the transmit path of the same port. *Figure 4-7* shows the signal paths when a remote loopback is activated on a data port connected to the SDH network.

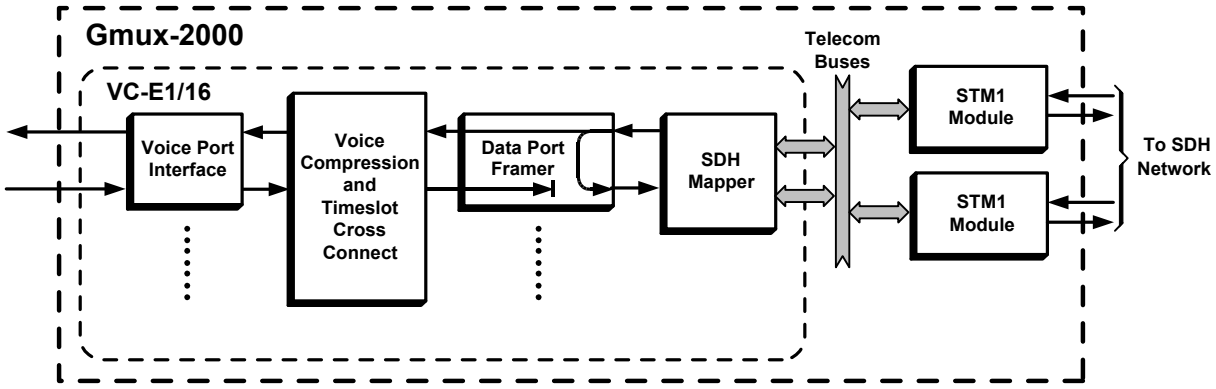



*Figure 4-7. Remote Loopback on Data Port Connected to SDH Network*

This test checks the whole transmission path between any remote user's equipment that connects through the network to the local data port, including the SDH transmission plant, the STM1 module, the internal Gmux-2000 paths connecting the user's equipment to the local VC-E1/16 data port, and the E1 framer of the VC-E1/16 data port.

In both configurations, the test signal is provided by the remote equipment connected to the local VC-E1/16 through the tested data port: any such equipment must receive its own transmission.

To ensure that the user's equipment connected to the tested port is capable of providing a good signal, the remote data port loopback should be activated on the local VC-E1/16 port only after checking that the remote user's equipment, and each equipment along the path, operates normally while its own local loopback is activated, and all the equipment along the path.

## *4.4* Troubleshooting Instructions

### Preliminary Actions

If the problem is detected the first time the module is put into operation, perform the following preliminary checks before proceeding:

Check for proper module installation and correct cable connections, in accordance with the system installation plan.

Check the module configuration parameters in accordance with the specific application requirements, as provided by the system administrator.

If the Gmux-2000 nodal (system) clock is to be locked to the clock recovered from one of the ports of the VC-E1/16 module, make sure a suitable fallback clock source is configured and provides a good clock signal.

### Troubleshooting Hardware Problems

In case a problem occurs, check the displayed alarm messages and refer to the Gmux-2000 Installation and Operation Manual for their interpretation.

If after collecting all the relevant information, the problem appears to be related to the operation of one of the VC-E1/16 ports, perform the actions listed below, until the problem is corrected:

1. Reset the VC-E1/16 and then read the results of the power-up test: if a technical problem is reported, replace the VC-E1/16.
2. If the power-up test is successfully ended, try to identify the type of port most likely to cause the problem:
   - A problem at a voice port affects only traffic carried by bundles serving that port.
   - A problem at a data port affects all the traffic passing through the port (multiple voice ports).

3. If the equipment connected to a local VC-E1/16 port reports loss of signal, activate the remote loopback on the corresponding port. If the problem disappears while the loopback is activated, replace the VC-E1/16 module.
4. You can rapidly check the link from a local port to the port at the remote Gmux-2000 unit by activating the local port loopback at the local VC-E1/16 port.

   If the test fails, there is a problem with the transmission through the network, or with the VC-E1/16 module. Repeat the test after carefully checking all the configuration parameters of the module and its ports. If the problem persists, replace the module and check again.

## Handling Service Problems

The main tool used to analyze unsatisfactory service, or service degradation, is to use the performance statistics and status data collection tools provided by the Gmux-2000. These tools are supplemented by an extensive set of dedicated performance monitoring functions for the VC-E1/16.

*Section 4.2* presents a detailed description of the VC-E1/16 performance monitoring functions, explains the various parameters, and provides guidelines for interpreting the displayed data.

By analyzing the performance statistics and status data, it is possible to identify the unsatisfactory performance area, and take steps to change the system configuration and enhance resources such a network bandwidth, etc., as required to improve the service quality. It is also recommended to check *Section 4.5* for answers to frequently asked questions.

Contact RAD Technical Support Department if you need additional help with your particular application.

# 4.5 Frequently Asked Questions

**Q**: A VC-E1/16 E1 port (either voice or data) reports remote loss of synchronization. What could be the possible causes of this condition?

**A:** Possible causes for a remote loss of synchronization alarm on a local E1 port:

- Incorrect cabling or bad physical connections causes loss of synchronization at the remote port directly connected to the local port.
- For a voice port: the destination port at the remote unit loses synchronization, and sends AIS signal towards the PBX, which also loses synchronization. In response, the PBX sends toward the local voice port a RAI indication.
- A bundle serving a whole port is out of service, thus causing the transmission of AIS signals to be transmitted from the relevant voice ports to the PBXs on both sides.
- The E1 framing mode of connected PBX/switch might be set to support CRC-4, while the local port is configured without CRC-4 support. This

causes the PBX/switch to signal to VC-E1/16 port that it is not synchronized.

**Q:** Can I use a VC-E1/16 module in a link with a Vmux-2100 port with T1 framing?

**A:** Yes.

VC-E1/16 enables you to connect between sites using E1-based voice traffic and sites using T1-based voice traffic. The only limitation is the total number of timeslots, which is maximum 24 (an E1-to-E1 link can transfer 30 timeslots).

Make sure to configure each side with appropriate parameters, in particular select the correct far-end parameters (for example, correct companding laws), and signaling profiles. If any additional conversions are required on the signaling protocol itself (for example, between different national variants), these should be performed externally.

**Q:** What is a BPV error, and what are the causes for such errors?

**A:** A BPV (bipolar polarity violation) error event is a line coding violation:

- For an AMI-coded signal, this is the occurrence of a pulse of the same polarity as the previous pulse.
- For a B8ZS- or HDB3-coded signal, this is the occurrence of a pulse of the same polarity as the previous pulse without being a part of the zero substitution code.

Such errors occur under marginal transmission conditions, which cause bit errors at the receiving side. For framing modes with CRC-4 support (**Framed MF – CRC** and **Framed – CRC**), you will see BER alarms; however, when the framing mode does not support CRC-4, only line coding errors can be detected.

**Q:** Can the bundle **Voice Coder** be changed remotely via Telnet? Will downtime occur while implementing this change?

**A:** Yes, the bundle **Voice Coder** can be changed remotely via Telnet. The bundle will be temporarily down until the change is completed at both ends. The management traffic is not affected by this change.

**Q:** Is it possible to use VC-E1/16 over low quality TDM connections (BER of $10^{-3}$ to $10^{-5}$)?

**A:** VC-E1/16 can use low quality TDM connections, but service quality may be degraded.

- The effect of marginal BER on the connection to a voice port is similar to the effect on other voice equipment.
- The effect on marginal BER on the network connection is more severe, because at the worst value allowed by the E1 transmission standards (BER of $10^{-3}$), it is quite possible that the bundle will go out-of-service from time to time (depending on the setting of the connectivity parameters). AIS will then be generated on both sides, until the bundle connectivity is re-established.

  The bundle may remain up, but some packets will be lost. In this case, the voice quality will be degraded. In addition, fax and modem transmission can be affected as well in case of packet loss.

Therefore, it is not possible to guarantee that in the worst-case scenario ($10^{-3}$) the VC-E1/16 will operate satisfactory.

**Q:** Does VC-E1/16 support ISDN data calls, such as WAP (V.110)?

**A:** The VC-E1/16 does not support ISDN data calls.

**Q:** Can the Signal-to-Noise ratio (SNR) be used to evaluate voice quality for VC-E1/16?

**A:** For compressed voice, the compression algorithm preserves intelligibility, but it is not possible to use SNR (which measures the difference between the signal power and the noise floor) to determine quality. You must use voice quality tests, such as MOS.

See also next question.

**Q:** Why do I get poor results when measuring voice quality over VC-E1/16 using PSQM?

**A:** PSQM (Perceptual Speech Quality Measurement) is not a suitable voice quality measurement technique for VoIP and voice compression systems that implement VAD (Voice Activity Detection) and Silence Suppression mechanisms. PSQM cannot synchronize between the reference and recorded files when there are gaps of silence, hence the poor results.

Acceptable voice quality techniques are PAMS (Perceptual Analysis Measurement System) and PESQ (Perceptual Evaluation of Speech Quality).

**Q:** Can the VC-E1/16 E1 ports provide an unbalanced 75 Ω interface terminated in BNC connectors?

**A:** VC-E1/16 E1 ports can be configured to provide balanced (120 Ω) or unbalanced 75 Ω interfaces, and therefore first select the unbalanced interface for the desired ports. After selecting the the unbalanced interface, use the following methods:

- For data ports, use the adapter cable CBL-RJ45/2BNC/E1/X, offered by RAD.
- For voice ports, use an open-ended cable, e.g.,, CBL-TELCO-OPEN/2M or CBL-TELCO-OPEN/10M, offered by RAD, to connect to a patch panel with BNC connectors.

In addition, conversion can also be achieved by use of an external balun converter (such as BE-1 from RAD).

**Q:** After activating a loopback on a VC-E1/16 data port, IP problems started to appear. What is the reason?

**A:** The loopback causes the IP traffic sent by the VC-E1/16 port to be looped back. This does not permit IP communication, and therefore problems are reported.

**Q:** Can a VC-E1/16 voice connection be tested using voice frequency test equipment?

**A:** Yes, but pay attention to the following considerations:

- Since VC-E1/16 compresses the voice traffic using very specialized algorithms, there is no guarantee that a sinusoidal tone can pass

undistorted, hence the transmitted test tone might suffer amplitude and/or phase distortions although voice can be satisfactorily transmitted.

- In any case, do not use tones with special functions, that are detected by VC-E1/16 relay mechanisms, for example:
  - 1100 Hz (CNG) will cause the VC-E1/16 to switch into fax-relay mode (if fax relay is enabled).
  - 2100 Hz (fax/modem CED) will cause the VC-E1/16 to switch into VBD (if modem relay is enabled).
  - 2000 Hz (COT - Continuity Test) will be relayed only if Custom Tone Detection is enabled.

  The above tones may be detected within a range of ±50 Hz around the nominal frequency (e.g. 1080 Hz can be detected as 1100 Hz). Therefore when testing voice frequency across VC-E1/16, you should avoid the following frequency ranges: 1050-1150 Hz, 1950-2150 Hz.

**Q:** Is it possible to send a BER test pattern across a VC-E1/16 voice channel?

**A:** No.

VC-E1/16 performs voice compression, not circuit emulation, and therefore it does not reconstruct at the remote end the original bit stream that was transmitted into the VC-E1/16 at the local end.

Moreover, the audio signal received by the local VC-E1/16 is compressed, and silence suppression techniques are also applied. The reverse process is carried out at the remote end, and the resulting regenerated audio signal is then transmitted in digital format (PCM) to the remote TDM equipment connected to the VC-E1/16 (PBX/switch). The output signal is a very good replica of the original input signal, but it is not represented by exactly the same PCM bytes, hence the resulting output bit stream differs from the original input data.

As a result of the above, it is not possible to perform BER test on an E1 stream that is compressed by VC-E1/16.

**Q:** How can I enable VC-E1/16 to support R2-MFC?

**A:** Perform the following steps to enable VC-E1/16 to support R2-MFC:

1. R2-MFC is a CAS protocol, and therefore, the E1 framing mode must be configured as **Framed-MF** (If the PBX uses CRC, **Framed MF- CRC4** should be used).
2. The VC-E1/16 DSP supports MFC relaying, and therefore at the other end of the link the MFC tones are regenerated by the DSP.

   By default, the MFC relay is disabled. Therefore when the VC-E1/16 must support MFC tones, enable MFRC relaying, and configure the required parameters, mainly **Minimum Pulse Width** and **Minimum Power Level**.

   Proper selection of these parameters prevents false detection of MFC tones (these tones appear within speech for short intervals). The result of this false detection is that the remote party will occasionally hear short MFC tones. In order to minimize the false detection of MFC tones, these two parameters are used to fine-tune the definition of the MFC tone shape. The goal is to lower the sensitivity of the DSP to MFC tones

in a way that only real MFC tones will be handled as MFC tones, thus avoiding false detection of speech tones (voice) as MFC tones. To lower this sensitivity, increase the MFC tone pulse width and lower its level.

However, if the false detection problem is not observed, it is recommended to leave the MFC tone pulse and level parameters at their default values.

**Q:** What is the end-to-end processing delay contributed by the VC-E1/16 when working with different compression algorithms (coders)?

**A:** The end-to-end processing delay contributed by the VC-E1/16 is as follows:

- G723.1A (5.3/6.4 kbps) – 120 to 150 msec (depending on packet interval)
- G729.A/8 kbps – 70 msec
- G.711 (A-law, 64 kbps) – 45 msec.

When used in an IP network, the VC-E1/16 uses jitter buffers in order to compensate for packet delay variation. The jitter buffers are implemented in the VC-E1/16 DSPs (Digital Signal Processors), and they are automatically adapted to the packet delay variation. The delay variation tolerance range is 10 to 300 msec.

Each timeslot has its own jitter buffer. The size of the VC-E1/16 jitter buffers is not user-configurable. The jitter buffer size is changed according to the packet delay variation, which is detected by the DSP. For this purpose, each voice packet sent out of the DSP includes a time stamp; the receiving DSP analyzes the time stamps on the received packets, and adjusts the size of the jitter buffer according to the delay between the received packets.

**Example**: When using G.723.1 compression, the DSP expects to receive a packet every 30 msec. If the delay between the received packets is higher than 30 msec, the DSP will increase the size of the jitter buffer so that data will not be lost.

**Q:** How does the VC-E1/16 inform the local PBX that a bundle is out-of-service?

**A:** When one of the VC-E1/16 bundles is in the out-of-service state (no response to the connectivity messages), the VC-E1/16 sends AIS to the local PBX.

In the CAS mode, if the E1 stream is split over more than one bundle, then the VC-E1/16 generates an OOS pattern on the signaling bits of that bundle (carried by the ABCD bits in timeslot 16).

The OOS pattern is determined by the signaling profile being used on the corresponding E1 voice port of the VC-E1/16 voice module.

**Q:** When using the Q.50 feature, how does VC-E1/16 inform the switches/PBXs on both sides to block new calls?

**A:** Every second, VC-E1/16 checks the actual bandwidth utilization on the network connection. When the specified Q.50 bandwidth utilization threshold is reached (user configurable, default 88%), the VC-E1/16 port inserts the appropriate ***block new calls*** signaling element in timeslot 16 in accordance with the selected ITU-T Rec. Q.50 protocol (Annex A or Annex B).

The time this indication reaches the local switches/PBXs is the response, which typically 100 to 200 msec. The time this indication reaches the remote switches/PBXs is longer, as it also depends on the network propagation delay. The network delay naturally depends on other factors (external to VC-E1/16).

**Q:** I used the bandwidth calculator to calculate the network connection bandwidth needed for my application, using an estimated silence percentage. What will happen if I my silence percentage estimation was too high, and as a consequence I will exceed the bandwidth over the VC-E1/16 E1 network connection?

**A:** The bandwidth calculator (included in the documentation CD) is a tool that helps you estimate the bandwidth required in a certain application/situation. Since it only provides you with an estimate, we recommend monitoring the silence percentage after actual deployment of the system, and recalculating the required bandwidth.

When the available bandwidth is exceeded, some of the packets are lost, and there will be some voice quality degradation. Activating the bandwidth control feature can minimize the impact on voice quality.

The frame size and the packetization interval affect the transmission throughput and the delay. Smaller frames and shorter packetization intervals will result in lower delay, but on the other hand will reduce throughput.

**Q:** Can double compression be prevented when passing through more than one compressed voice hop?

**A:** When voice travels through several VC-E1/16 and Vmux-2100 gateways, the intermediate units sense that the voice data has already been compressed, and do not uncompress and recompress the data as it moves through the phone network. This ensures that voice is only compressed once (at the originating VC-E1/16 or Vmux-2100 gateway) and decompressed once (at the terminating VC-E1/16 or Vmux-2100 gateway), resulting in the same quality as one hop.

How is this done? The VC-E1/16 continuously transmits a very low frequency signal (38 Hz) that enables any other VC-E1/16 or Vmux-2100 receiving this information to detect that it is coming from another VC-E1/16 or Vmux-2100. This 38 Hz tone cannot pass through analog telephony equipment (which is usually limited to 300 to 3400 Hz), but it does pass through digital PBXs (as long as there are no digital-to-analog conversions in-between).

Once VC-E1/16 or Vmux-2100 units detect each other, they both enter the **Super Tandem** mode, and immediately stop compressing/decompressing on this segment.

For this mechanism to work, make sure to enable the **Super Tandem** mode for the relevant bundles.

**Q:** What is the difference between **HDLC** and **SS7** timeslot types?

**A:** The **SS7** traffic type was added to the existing **HDLC** type to support both ISDN-PRI and SS7 CCS signaling modes.

This was necessary because when passing SS7 traffic, you must set a FISU suppression rate, which affects the entire VC-E1/16. This makes it impossible to run ISDN PRI as well as SS7 on the same VC-E1/16 (even on different E1 ports), as the PRI frames get corrupted due to the FISU suppression.

**The proper procedure is:**

1. Select **SS7** for SS7 traffic
2. Select **HDLC** for non-SS7 traffic (e.g. ISDN PRI)

**Q:** How many voice bundles can be configured for one VC-E1/16 unit?

**A:** The VC-E1/16 can support up to 32 active bundles.

**Q:** Does the VC-E1/16 support ITU-T Rec. Q.50?

**A:** VC-E1/16 supports both Annex A and Annex B of the ITU-T Rec. Q.50 protocol, but only for the 3.1 kHz audio or speech connection type.

Annex A also includes signals which are transported over three spare bits (5, 7, and 8) of timeslot 16 multiframes. These are used to transmit the transmission resource information and maintenance signals when it is possible to carry timeslot 16 frame 0 between the DCME (Digital Circuit Multiplication Equipment – in this case the VC-E1/16) and the International Switching Center. Note that any equipment placed between the DCME and the International Switching Center, such as timeslot interchange equipment, may prevent this transmission.

Q: Can VC-E1/16 transmit fax over links with a long delay?

A: Test results show that even with a delay of 900 msec in one direction, there are no problem sending/receiving fax messages. Therefore, according to this test, it will be possible to use VC-E1/16 in links that introduce long delays (usually a satellite application), and to be able to support a fax server application.

Q: How does the VC-E1/16 handle fax or modem transmission?

A: The DSP that is used by the VC-E1/16 voice modules has a fax/modem discriminator that allows it to detect and distinguish modem and fax start signals, according to ITU-T standards.

The VC-E1/16 DSP has a fax relay. When the DSP (Digital Signaling Processor) detects fax transmission over the voice channel, it stops voice compression and transfers the fax transmission at a user-configurable rate (up to 14.4 kbps). When in fax mode, the DSP serves in fact just as a fax machine: it demodulates the analog signal at the receiving end, passes the resulting digital information to the remote end, and then remodulates the digital info into an analog signal, simulating the original fax machine.

Supported fax standards are: G3 fax per ITU-T Rec. T.30.

When modem transmissions can be handled in a similar way. Supported modem standards are: ITU-T Rec. V.34 (33.6 kbps), V.32bis (14.4 kbps), V.22bis (4.8 kbps).

## 4.6 Technical Support

Technical support for this product can be obtained from the local distributor from whom it was purchased.

For further information, please contact the RAD distributor nearest you or one of RAD's offices worldwide. This information can be found at www.rad.com (**offices – About RAD > Worldwide Offices; distributors – Where to Buy > End Users**).

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# Customer Response Form

RAD Data Communications would like your help in improving its product documentation. Please complete and return this form by mail or by fax or send us an e-mail with your comments.

Thank you for your assistance!

**Manual Name:** VC-E1/16, VC-E1/12

**Publication Number:** 358-209-07/08

Please grade the manual according to the following factors:

| | *Excellent* | *Good* | *Fair* | *Poor* | *Very Poor* |
|---|---|---|---|---|---|
| Installation instructions | ❒ | ❒ | ❒ | ❒ | ❒ |
| Operating instructions | ❒ | ❒ | ❒ | ❒ | ❒ |
| Manual organization | ❒ | ❒ | ❒ | ❒ | ❒ |
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What did you like about the manual?

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## Error Report

Type of error(s) or problem(s):

- ❒ Incompatibility with product
- ❒ Difficulty in understanding text
- ❒ Regulatory information (Safety, Compliance, Warnings, etc.)
- ❒ Difficulty in finding needed information
- ❒ Missing information
- ❒ Illogical flow of information
- ❒ Style (spelling, grammar, references, etc.)
- ❒ Appearance
- ❒ Other _____

Please list the exact page numbers with the error(s), detail the errors you found (information missing, unclear or inadequately explained, etc.) and attach the page to your fax, if necessary.

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Address: _____

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